MultiVOIP™

Voice / Fax over IP Networks

User Guide for Voice/IP Gateways

Digital Models (T1, E1, ISDN-PRI):
MVP-2410/3010

MultiTech® Systems

User Guide

S000384A

Digital MultiVOIP Units (Models MVP2410, MVP3010)
Upgrade Units (MVP24-48 and MVP30-60)

Record of Revisions

Revision	Description
A	Doc re-organization. Follows S000249K. (09/26/05) Describes 4.08 software release.

Patents

This Product is covered by one or more of the following U.S. Patent Numbers: ***6151333, 5757801, 5682386**, **5.301.274; 5.309.562**; **5.355.365; 5.355.653; 5.452.289; 5.453.986***. Other Patents Pending.

Trademark

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Chapter 1: Overview

About This Manual

This manual is about Voice-over-IP products made by Multi-Tech Systems, Inc. It describes three analog MultiVOIP units, models MVP810, MVP410, and MVP210.

These MultiVOIP units can inter-operate with other contemporary analog MultiVOIP units (MVP130 & MVP130FXS), with contemporary BRI MultiVOIP units (MVP410ST & MVP810ST), with contemporary digital T1/E1/ISDN-PRI MultiVOIP units (MVP2410 and MVP3010), and with the earlier generation of MultiVOIP products (MVP200, MVP400, MVP800, MVP120, etc.)

The table below (on next page) describes the vital characteristics of the various models described in this manual.

How to Use This Manual. ***In short, use the index and the examples.*** When our readers crack open this large manual, they generally need one of two things: information on a very specific software setting or technical parameter (about telephony or IP) *or* they need help when setting up phonebooks for their voip systems. The index gives quick access to voip settings and parameters. It's detailed. Use it. The best way to learn about phonebooks is to wade through examples like those in our chapters on T1 (North American standard) Phonebooks and E1 (Euro standard) Phonebooks. Finally, this manual is meant to be comprehensive. If you notice that something important is lacking, please let us know.

Additional Resources. The MultiTech web site (www.multitech.com) offers both a list of Frequently Asked Questions (the MultiVOIP FAQ) and a collection of resolutions of issues that MultiVOIP users have encountered (these are Troubleshooting Resolutions in the searchable Knowledge Base).

MultiVOIP Product Family					
Description / Model		MVP-2410	MVP 24-48	MVP 3010	MVP 30-60
Function		T1 digital VOIP unit	T1 digital VOIP add-on card	E1 digital VOIP unit	E1 digital VOIP add-on card
Capacity		24 channels	24 added channels	30 channels	30 added channels
Chassis/ Mounting		19” 1U rack mount	circuit card only	19” 1U rack mount	circuit card only
Description / Model	MVP 810	MVP 428	MVP 410	MVP 210	MVP-130/ 130FXS
Function	analog voip	add-on card	analog voip	analog voip	analog voip
Capacity	8 channels	4 added channels	4 channels	2 channels	1 channel
Chassis/ Mounting	19” 1U rack mount	circuit card only	19” 1U rack mount	Table top	table top
Description / Model	MVP810ST		MVP410ST		
Function	ISDN-BRI voip		ISDN-BRI voip		
Capacity	4 ISDN lines (8 B-channels)		2 ISDN lines (4 B-channels)		
Chassis/ Mounting	19” 1U rack mount		19” 1U rack mount		
	1. “BRI” means Basic Rate Interface.				

Introduction to TI MultiVOIPs (MVP2410 & MVP24-48)

We proudly present MultiTech's T1 Digital Multi-VOIP products. The MVP2410 is a rack-mount model; and the MVP24-48 is an add-on expansion card that doubles the capacity of the MVP2410 without adding another chassis. These voice-over-IP products have fax capabilities. These models adhere to the North American standard of T1 trunk telephony using digital 24-channel time-division multiplexing, which allows 24 phone conversations to occur on the T1 line simultaneously. They can also accommodate T1 lines of the ISDN Primary Rate Interface type (ISDN-PRI).

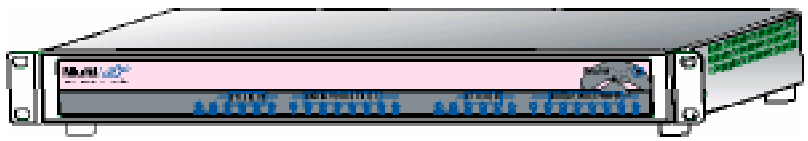


Figure 1-1. MultiVOIP MVP2410 LEDs

Scale-ability. The MVP2410 is tailored to companies needing more than a few voice-over-IP lines, but not needing carrier-class equipment. When expansion is needed, the MVP2410 can be field-upgraded into a dual T1 unit by installing the MVP24-48 kit, which is essentially a second MultiVOIP motherboard that fits in an open expansion-card slot in the MVP2410. The upgraded dual unit then accommodates two T1 lines.

T1 VOIP Traffic. The MVP2410 accepts its outbound traffic from a T1 trunk that's connected to either a PBX or to a telco/carrier. The MVP2410 transforms the telephony signals into IP packets for transmission on LANs, WANs, or the Internet. Inbound IP data traffic is converted to telephony data and signaling.

When connected to PBX. When connected to a PBX, the MVP2410 creates a network node served by 10/100-Base T connections. Local PBX phone extensions gain toll-free access to all phone stations directly connected to the VOIP network. Phone extensions at any VOIP location also gain toll-free access to the entire local public-switched telephone network (PSTN) at every other VOIP location in the system.

When connected to PSTN. When the T1 line(s) connected to the MVP2410 are connected directly to the PSTN, the unit becomes a Point-of-Presence server dedicated to local calls off-net.

H.323, SIP & SPP. Being H.323 compatible, the MVP2410 can place calls to telephone equipment at remote IP network locations that also contain H.323 compatible voice-over-IP gateways. It will interface with H.323 software and H.323 gatekeeper units. H.323 specifications also bring to voip telephony many special features common to conventional telephony. H.323 features of this kind that have been implemented into the MultiVOIP include Call Hold, Call Waiting, Call Name Identification, Call Forwarding (from the H.450 standard), and Call Transfer (H.450.2 from H.323 Version 2). The fourth version of the H.323 standard improves system resource usage (esp. logical port or socket usage) by handling call signaling more compactly and allowing use of the low-overhead UDP protocol instead of the error-correcting TCP protocol where possible.

The MultiVOIP is also SIP-compatible. (“SIP” means Session Initiation Protocol.) However, H.450 Supplementary Services features can be used under H.323 only and not under SIP.

SPP (Single-Port Protocol) is a non-standard protocol developed by Multi-Tech. SPP is not compatible with the “Proprietary” protocol used in Multi-Tech’s earlier generation of voip gateways. SPP offers advantages in certain situations, especially when firewalls are used and when dynamic IP address assignment is needed. However, when SPP is used, certain features of SIP and H.323 will not be available and SPP will not inter-operate with voip systems using H.323 or SIP.

Data Compression & Quality of Service. The MultiVOIP MVP2410 comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

VOIP Functions. The MultiVOIP MVP2410 gateway performs four basic functions: (a) it converts a dialed number into an IP address, (b) it sends voice over the data network, (c) it establishes a connection with another VOIP gateway at a remote site, and (d) it receives voice over the data network. Voice is handled as IP packets with a variety of compression options. Each T1 connection to the MultiVOIP provides 24 time-slot channels to connect to the telco or to serve phone or fax stations connected to a PBX.

Ports. The MVP2410 has one 10/100 Mbps Ethernet LAN interface and one Command port for configuration. An MVP2410 upgraded with the MVP24-48 kit will have two Ethernet LAN interfaces and two Command ports.

PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

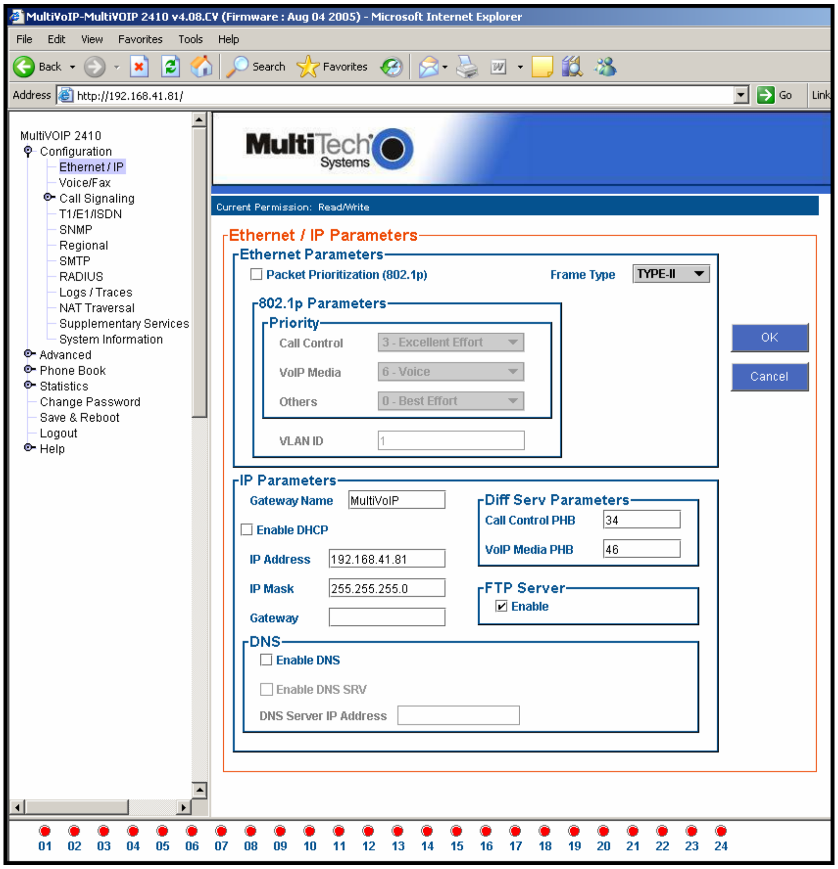
RADIUS Support. Inter-operation with a RADIUS server allows for call accounting (especially for billing) on a voip system. The MultiVOIP supports inter-operation with RADIUS servers for the RADIUS accounting function (but not the RADIUS authentication function).

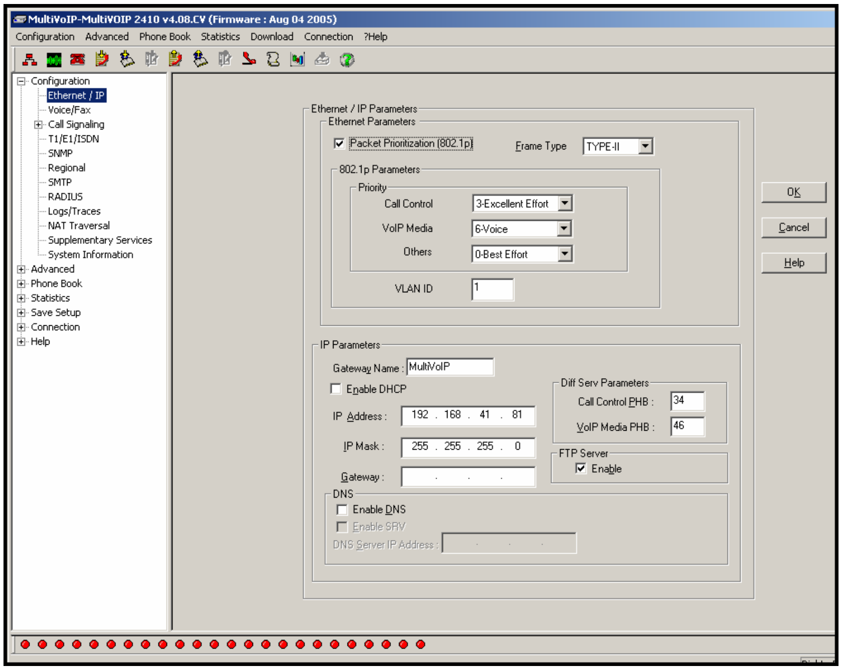
STUN Support. The STUN protocol (Simple Traversal of UDP through NATs (Network Address Translation)) assists with the packet routing functions of devices behind NAT firewalls or routers. The MultiVOIP supports inter-operation with STUN servers and NATs (SIP based environment only).

Gatekeeper. T1 voip systems can have gatekeeper functionality by adding, as an endpoint, a Multi-Tech standalone gatekeeper (special software residing in separate hardware). Gatekeepers are optional but useful within voip systems. The gatekeeper acts as the 'clearinghouse' for all calls within its zone. MultiTech's stand-alone gatekeeper software performs all of the standard gatekeepers functions (address translation, admission control, and bandwidth control) and also supports many valuable optional functions (call control signaling, call authorization, bandwidth management, and call management).

Management. Configuration and system management can be done locally with the MultiVOIP configuration software. After an IP address has been assigned locally, other configuration can be done remotely using the MultiVOIP web browser GUI. Remote system management can be done with the MultiVoipManager SNMP software or via the MultiVOIP web browser GUI. All of these control software packages are included on the Product CD.

While the web GUI's appearance differs slightly, its content and organization are essentially the same as that of the Windows GUI (except for logging).

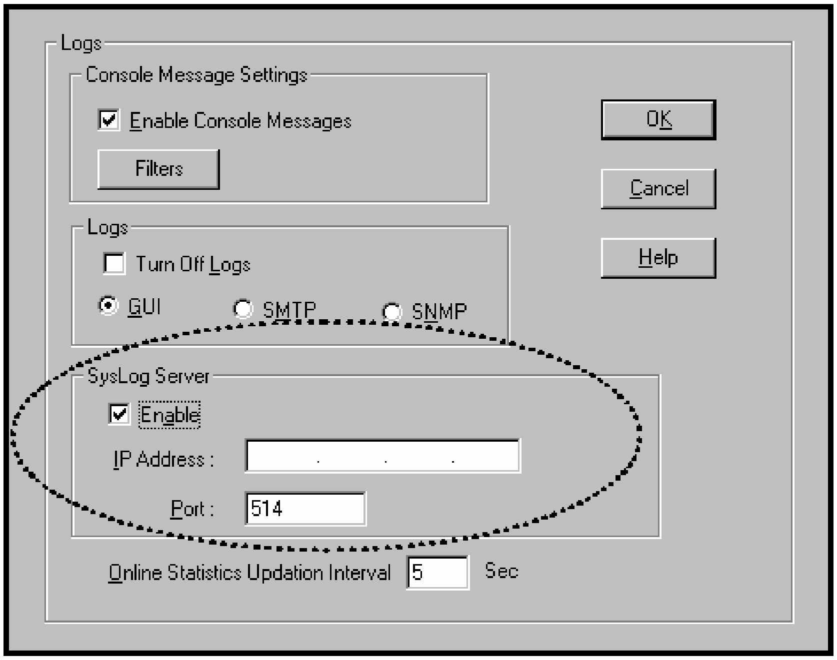




The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

Once you've begun using the web browser GUI, you can go back to the MultiVOIP Windows GUI at any time. However, you must log out of the web browser GUI before using the MultiVOIP Windows GUI.

Logging of System Events. MultiTech has built SysLog Server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.



The SysLog Server resides in the MultiVOIP unit itself. To implement this functionality, you will need a SysLog client program (sometimes referred to as a "daemon"). SysLog client programs, both paid and freeware, can be obtained from Kiwi Enterprises, among other firms. See www.kiwisyslog.com. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

MultiTech Systems does not endorse any particular SysLog client program. SysLog client programs by any qualified provider should suffice for use with MultiVOIP units. Kiwi's brief description of their SysLog program indicates the typical scope of such programs. "Kiwi Syslog Daemon is a freeware Syslog Daemon for the Windows platform. It receives, logs, displays and forwards Syslog messages from hosts such as routers, switches, Unix hosts and any other syslog enabled device. There are many customizable options available."

Supplementary Telephony Services. The H.450 standard (an addition to H.323) brings to voip telephony more of the premium features found in PSTN and PBX telephony. MultiVOIP units offer five of these H.450 features: Call Transfer, Call Hold, Call Waiting, Call Name Identification (not the same as Caller ID), and Call Forwarding. (The first four features are found in the "Supplementary Services" window; the fifth, Call Forwarding, appears in the Add/Edit Inbound phonebook screen.) Note that the first three features are closely related. All of these H.450 features are supported for H.323 operation only; they are *not* supported for SIP or SPP.

T1 Front Panel LEDs

The MVP2410 and MVP24-48 both use a common main circuit board or motherboard. Consequently the LED indicators are the same for both.

Active LEDs. The MVP2410 front panel has two sets of identical LEDs. In the MVP2410 as shipped (that is, without an expansion card), the left-hand set of LEDs is functional whereas the right-hand set is not.

When the MVP2410 has been upgraded with an MVP24-48 kit, the right-hand set of LEDs will also become active.

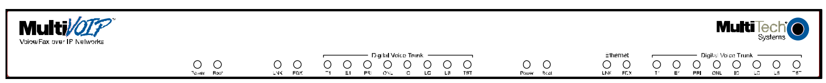


Figure 1-2: MVP2410 LEDs

T1 LED Descriptions. The descriptions below apply to the digital T1 MultiVOIP units. The MVP2410 has four sets of LEDs plus a lone LED at its far right end. As viewed from the front of the MVP2410, it is the two left groups that are active and present feedback about the operation of the unit. If an MVP24-48 expansion card is added to the MVP2410, the two LED groups on the right become operational with respect to the second T1 connection.

MVP2410 Front Panel LED Definitions	
LED NAME	**DESCRIPTION**
Power	Indicates presence of power.
Boot	After power up, the Boot LED will be on for about 10 seconds while the MVP2410 is booting.
FDX	Full-Duplex & Collision LED. This LED indicates whether the Ethernet connection is half-duplex or full-duplex (FDX) and, in half-duplex mode, indicates occurrence of data collisions. LED is on constantly for full-duplex mode; LED is off constantly for half-duplex mode. When operating in half-duplex mode, the LED will flash during data collisions.
LNK	Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While link is up, this LED will flash off to indicate data activity.
T1	When lit, indicates presence of T1 connection.
E1	E1. Not supported.
PRI	PRI. On if T1 line is of ISDN-Primary-Rate type.
ONL	Online. This LED is on when frame synchronization has been established on the T1/E1 link.
IC	IC LED is on when Internal Clocking is selected in T1/E1 configuration.
LC	Indicates Loss of Carrier.
LS	Indicates Loss of Signal.
Test	For testing purposes only.

Introduction to EI MultiVOIPs (MVP3010 & MVP30-60)

We proudly present MultiTech's E1 Digital Multi-VOIP products. The MVP3010 is a rack-mount model and the MVP30-60 is an add-on expansion card that doubles the capacity of the MVP3010 without adding another chassis. All of these voice-over-IP products have fax capabilities. All adhere to the European standard of E1 trunk telephony using digital 30-channel time-division multiplexing, which allows 30 phone conversations to occur on the E1 line simultaneously. All can also accommodate E1 lines of the ISDN Primary Rate Interface type (ISDN-PRI).

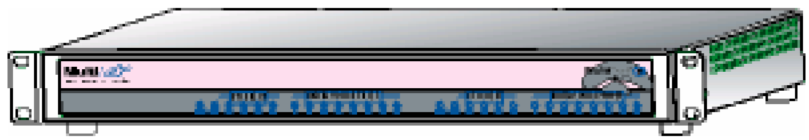


Figure 1-3. MultiVOIP MVP3010 Chassis

Scale-ability. The MVP3010 is tailored to companies needing more than a few voice-over-IP lines, but not needing carrier-class equipment. When expansion is needed, the MVP3010 can be field-upgraded into a dual E1 unit by installing the MVP30-60 kit, which is essentially a second MultiVOIP motherboard that fits into an open expansion-card slot in the MVP3010. The upgraded dual unit then accommodates two E1 lines.

E1 VOIP Traffic. The MVP3010 accepts its outbound traffic from an E1 trunk that's connected to either a PBX or to a telco/carrier. The MVP3010 transforms the telephony signals into IP packets for transmission on LANs, WANs, or the Internet. Inbound IP data traffic is converted to telephony data and signaling.

When connected to PBX. When connected to a PBX, the MVP3010 creates a network node served by 10/100-Base T connections. Local PBX phone extensions gain toll-free access to all phone stations directly connected to the VOIP network. Phone extensions at any VOIP location also gain local-rate access to the entire local public-switched telephone network (PSTN) at every other VOIP location in the system.

When connected to PSTN. When the E1 line(s) connected to the MVP3010 are connected directly to the PSTN, the unit becomes a Point-of-Presence server dedicated to local calls off-net.

H. 323, SIP, & SPP. Being H.323 compatible, the MVP3010 can place calls to telephone equipment at remote IP network locations that also contain H.323 compatible voice-over-IP gateways. It will interface with H.323 software and H.323 gatekeeper units. H.323 specifications also bring to voip telephony many special features common to conventional telephony. H.323 features of this kind that have been implemented into the MultiVOIP include Call Hold, Call Waiting, Call Identification, Call Forwarding (from the H.450 standard), and Call Transfer (H.450.2 from H.323 Version 2). The fourth version of the H.323 standard improves system resource usage (esp. logical port or socket usage) by handling call signaling more compactly and allowing use of the low-overhead UDP protocol instead of the error-correcting TCP protocol where possible.

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Data Compression & Quality of Service. The MultiVOIP3010 comes equipped with a variety of data compression capabilities, including G.723, G.729, and G.711 and features DiffServ quality-of-service (QoS) capabilities.

VOIP Functions. The MultiVOIP MVP3010 gateway performs four basic functions: (a) it converts a dialed number into an IP address, (b) it sends voice over the data network, (c) it establishes a connection with another VOIP gateway at a remote site, and (d) it receives voice over the data network. Voice is handled as IP packets with a variety of compression options. Each E1 connection to the MultiVOIP provides 30 time-slot channels to connect to the telco or to serve phone or fax stations connected to a PBX.

Ports. The MVP3010 also has a 10/100 Mbps Ethernet LAN interface, and a Command port for configuration. An MVP3010 upgraded with the MVP30-60 kit will have two Ethernet LAN interfaces and two Command ports.

PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

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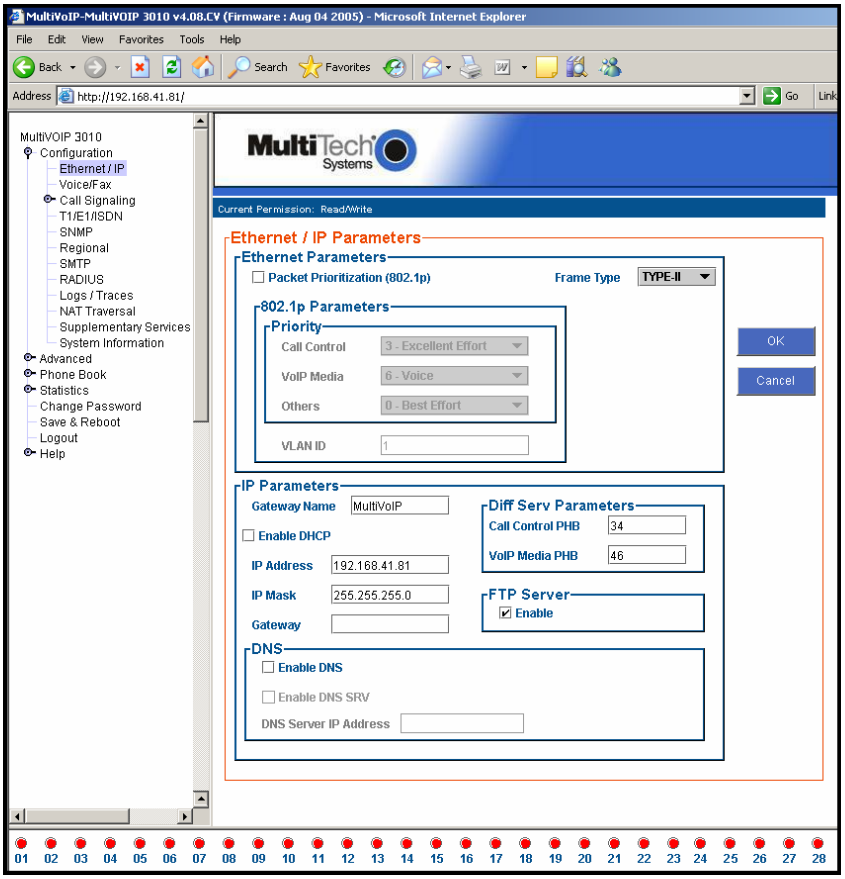

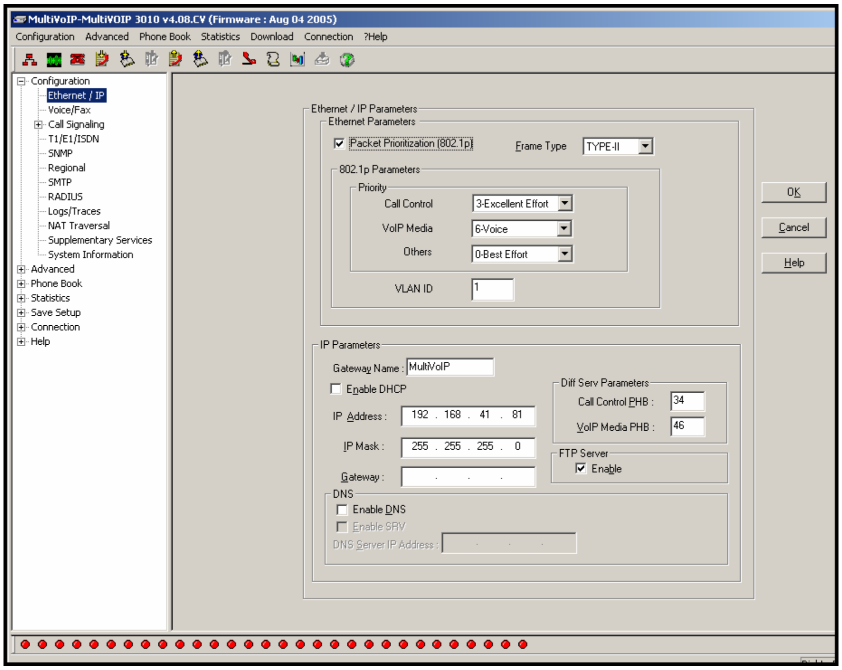

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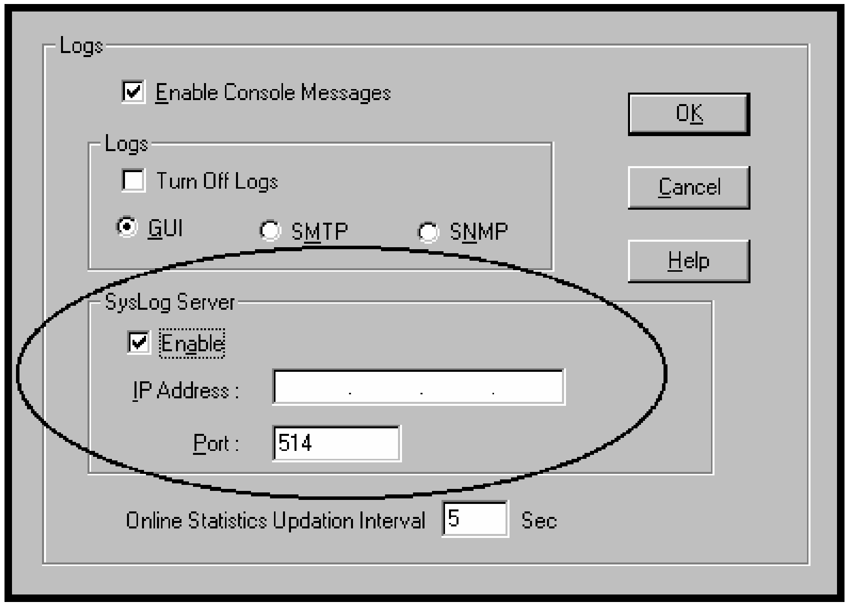

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E1 Front Panel LEDs

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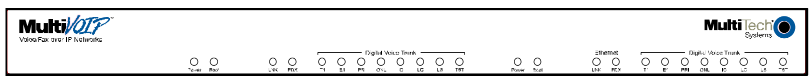


Figure 1-4: MVP3010 LEDs

Active LEDs. The MVP3010 front panel has two sets of identical LEDs. In the MVP3010 as shipped (that is, without an expansion card), the left-hand set of LEDs is functional whereas the right-hand set is not.

When the MVP3010 has been upgraded with an MVP30-60 kit, the right-hand set of LEDs will also become active.

E1 LED Descriptions

MVP3010 Front Panel LED Definitions	
LED NAME	**DESCRIPTION**
Power	Indicates presence of power.
Boot	After power up, the Boot LED will be on for about 10 seconds while the MVP3010 is booting.
FDX	Full-Duplex & Collision LED. This LED indicates whether the Ethernet connection is half-duplex or full-duplex (FDX) and, in half-duplex mode, indicates occurrence of data collisions. LED is on constantly for full-duplex mode; LED is off constantly for half-duplex mode. When operating in half-duplex mode, the LED will flash during data collisions.
LNK	Link/Activity LED. This LED is lit if Ethernet connection has been made. It is off when the link is down (i.e., when no Ethernet connection exists). While link is up, this LED will flash off to indicate data activity.
T1	T1. Not supported.
E1	E1. When lit, indicates presence of E1 connection.
PRI	PRI. On if E1 line is of ISDN-Primary-Rate type.
ONL	Online. This LED is on when frame synchronization has been established on the T1/E1 link.
IC	IC LED is on when Internal Clocking is selected in T1/E1 configuration.
LC	Indicates Loss of Carrier.
LS	Indicates Loss of Signal.
Test	For testing purposes only.

Specifications

Specs for Digital T1 MultiVOIP Units

Digital T1 MultiVOIP Specifications		
Parameter ……/Model	**MVP-2410**	**MVP-2410 w/ MVP24-48 Expansion Card**
Operating Voltage/Current	100-240 VAC 1.2 - 0.6 A	100-240 VAC 1.2 - 0.6 A
Mains Frequencies	50/60 Hz	50/60 Hz
Power Consumption	17 watts	27 watts
Mechanical Dimensions	1.75”H x 17.4”W x 8.75”D 4.5cm H x 44.2 cm W x 22.2 cm D	1.75”H x 17.4”W x 8.75”D 4.5cm H x 44.2 cm W x 22.2 cm D
Weight	7.1 lbs. (3.2 kg)	7.5 lbs. (3.4 kg)

Specs for Digital E1 MultiVOIP Units

Digital E1 MultiVOIP Specifications		
Parameter ……/Model	**MVP-3010**	**MVP-3010 w/ MVP30-60 Expansion Card**
Operating Voltage/Current	100-240 VAC 1.2 - 0.6 A	100-240 VAC 1.2 - 0.6 A
Mains Frequencies	50/60 Hz	50/60 Hz
Power Consumption	17 watts	27 watts
Mechanical Dimensions	1.75”H x 17.4”W x 8.75”D 4.5cm H x 44.2 cm W x 22.2 cm D	1.75”H x 17.4”W x 8.75”D 4.5cm H x 44.2 cm W x 22.2 cm D
Weight	7.1 lbs. (3.2 kg)	7.5 lbs. (3.4 kg)

Installation at a Glance

The basic steps of installing your MultiVOIP network involve unpacking the units, connecting the cables, and configuring the units using management software (MultiVOIP Configuration software) and confirming connectivity with another voip site. This process results in a fully functional Voice-Over-IP network.

Related Documentation

The MultiVOIP User Guide (the document you are now reading) comes in electronic form and is included on your system CD. It presents in-depth information on the features and functionality of Multi-Tech's MultiVOIP Product Family.

The CD media is produced using Adobe Acrobat™ for viewing and printing the user guide. To view or print your copy of a user guide, load Acrobat Reader™ on your system. The Acrobat Reader is included on the MultiVOIP CD and is also a free download from Adobe's Web Site:

www.adobe.com/prodindex/acrobat/readstep.html

This MultiVOIP User Guide is also available on Multi-Tech's Web site at:

http://www.multitech.com

Viewing and printing a user guide from the Web also requires that you have the Acrobat Reader loaded on your system. To select the MultiVOIP User Guide from the Multi-Tech Systems home page, click **Documents** and then click **MultiVOIP Family** in the product list drop-down window. All documents for this MultiVOIP Product Family will be displayed. You can then choose *User Guide (MultiVOIP Product Family)* to view or download the **.pdf** file.

Entries (organized by model number) in the "knowledge base" and 'troubleshooting resolutions' sections of the MultiTech web site (found under "Support") constitute another source of help for problems encountered in the field.

Chapter 2: Quick Start Instructions

The Quick Start Guide is a separate manual with streamlined instructions to get the MultiVOIP up and running quickly. These start-up instructions include assistance on setting up the MultiVOIP's Inbound and Outbound Phonebooks. These sections of the Quick Start Guide may be particularly useful for phonebook configuration:

Phonebook Starter Configuration
Phonebook Tips
Phonebook Example (One Common Situation)

The Quick Start Guide also contains a "Phonebook Worksheet" section. You may want to print out several worksheet copies. Paper copies can be very helpful in comparing phonebooks at multiple sites at a glance. This will assist you in making the phonebooks clear and consistent and will reduce 'surfing' between screens on the configuration program.

A printed Quick Start Guide is shipped with the MultiVOIP and an electronic copy is included on the Product CD.

Chapter 3: Mechanical Installation and Cabling

Introduction

When the MVP2410 or MVP3010 unit is to be installed into a rack, two able-bodied persons should participate.

Please read the safety notices before beginning installation.

Safety Warnings

Lithium Battery Caution

A lithium battery on the voice/fax channel board provides backup power for the timekeeping capability. The battery has an estimated life expectancy of ten years.

When the battery starts to weaken, the date and time may be incorrect. If the battery fails, the board must be sent back to Multi-Tech Systems for battery replacement.

Warning: There is danger of explosion if the battery is incorrectly replaced.

Safety Warnings Telecom

1. Never install telephone wiring during a lightning storm.
2. Never install a telephone jack in wet locations unless the jack is specifically designed for wet locations.
3. This product is to be used with UL and UL listed computers.
4. Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
5. Use caution when installing or modifying telephone lines.
6. Avoid using a telephone (other than a cordless type) during an electrical storm. There may be a remote risk of electrical shock from lightning.
7. Do not use a telephone in the vicinity of a gas leak.
8. To reduce the risk of fire, use only a UL-listed 26 AWG or larger telecommunication line cord.

Unpacking Your MultiVOIP

When unpacking your MultiVOIP, check to see that all of the items shown are included in the box. If any box contents are missing, contact MultiTech Tech Support at 1-800-972-2439.

Unpacking the MVP2410/3010

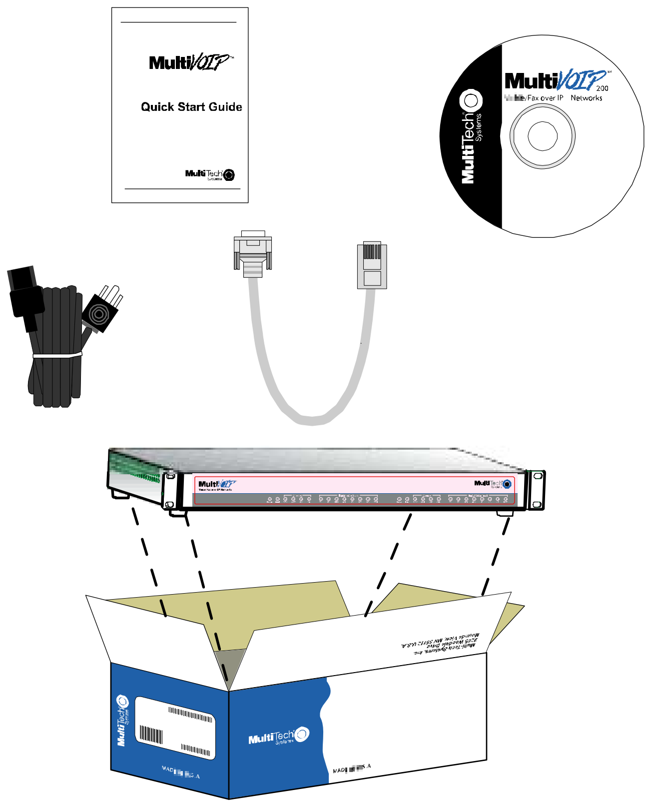


Figure 3-1: Unpacking the MVP2410/3010

Rack Mounting Instructions

The MultiVOIPs can be mounted in an industry-standard EIA 19-inch rack enclosure, as shown in Figure 3-2.

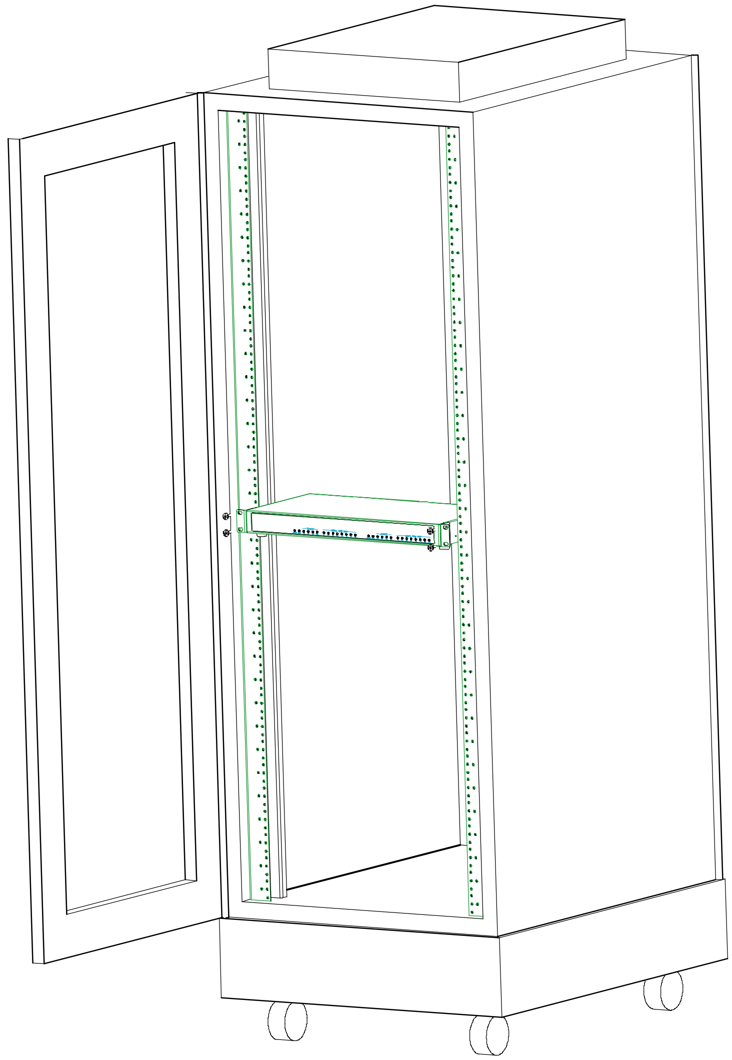


Figure 3-2: Rack-Mounting

Safety Recommendations for Rack Installations

Ensure proper installation of the unit in a closed or multi-unit enclosure by following the recommended installation as defined by the enclosure manufacturer. Do not place the unit directly on top of other equipment or place other equipment directly on top of the unit. If installing the unit in a closed or multi-unit enclosure, ensure adequate airflow within the rack so that the maximum recommended ambient temperature is not exceeded. Ensure that the unit is properly connected to earth ground by verifying that it is reliably grounded when mounted within a rack. If a power strip is used, ensure that the power strip provides adequate grounding of the attached apparatus.

When mounting the equipment in the rack, make sure mechanical loading is even to avoid a hazardous condition, such as loading heavy equipment in rack unevenly. The rack used should safely support the combined weight of all the equipment it supports.

Ensure that the mains supply circuit is capable of handling the load of the equipment. See the power label on the equipment for load requirements (full specifications for MultiVOIP models are presented in chapter 1 of this manual).

Maximum ambient temperature for the unit is 60 degrees Celsius (140 degrees Fahrenheit) at 20-90% non-condensing relative humidity. This equipment should only be installed by properly qualified service personnel. Only connect like circuits. In other words, connect SELV (Secondary Extra Low Voltage) circuits to SELV circuits and TN (Telecommunications Network) circuits to TN circuits.

19-Inch Rack Enclosure Mounting Procedure

Attaching the MultiVOIP to a rack-rail of an EIA 19-inch rack enclosure will certainly require two persons. Essentially, the technicians must attach the brackets to the MultiVOIP chassis with the screws provided, as shown in Figure 3-3, and then secure unit to rack rails by the brackets, as shown in Figure 3-4. Because equipment racks vary, screws for rack-rail mounting are not provided. Follow the instructions of the rack manufacturer and use screws that fit.

1. Position the right rack-mounting bracket on the MultiVOIP using the two vertical mounting screw holes.
2. Secure the bracket to the MultiVOIP using the two screws provided.
3. Position the left rack-mounting bracket on the MultiVOIP using the two vertical mounting screw holes.
4. Secure the bracket to the MultiVOIP using the two screws provided.
5. Remove feet (4) from the MultiVOIP unit.
6. Mount the MultiVOIP in the rack enclosure per the rack manufacture’s mounting procedure.

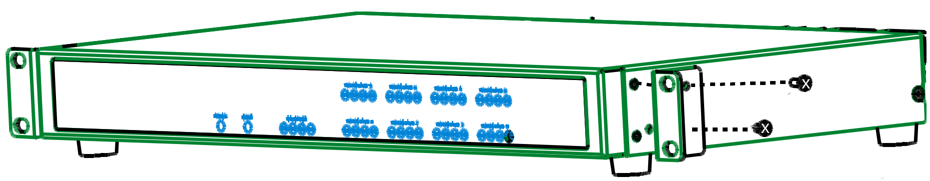


Figure 3-3: Bracket Attachment for Rack Mounting

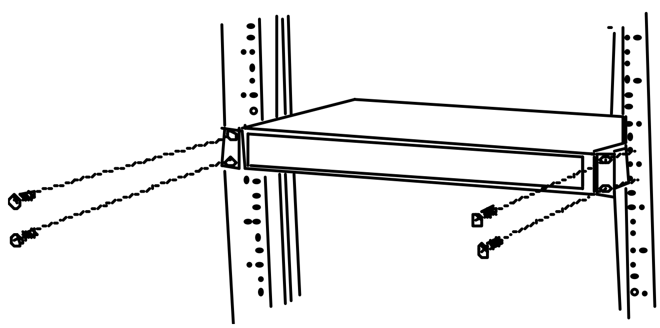


Figure 3-4: Attaching MultiVOIP to Rack Rail

Cabling

Cabling Procedure

Cabling your MultiVOIP entails making the proper connections for power, command port, phone system (T1/E1 line connected to PBX or telco office), and Ethernet network. Figure 3-5 shows the back panel connectors and the associated cable connections. The following procedure details the steps necessary for cabling your MultiVOIP.

1. Connect the power cord to a live AC outlet, then connect it to the MultiVOIP's power receptacle shown at top right in Figure 3-5.

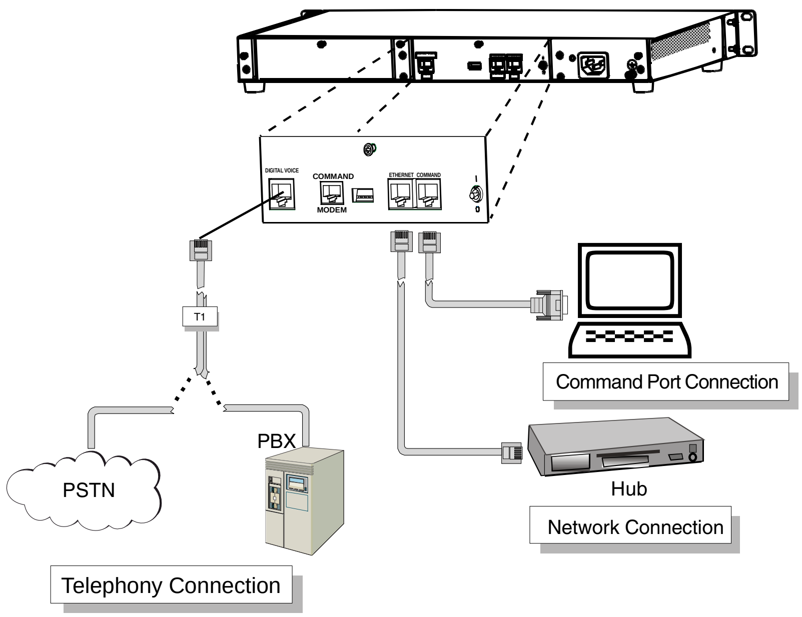


Figure 3-5. Cabling for MVP2410/3010

2. Connect the MultiVOIP to the PC (the computer that will hold the MultiVOIP software) using the RJ-45 to DB9 (female) cable provided with your unit. Plug the RJ-45 end of the cable into the **Command** port of the MultiVOIP and connect the other end (the DB9 connector) to the PC serial port you are using (typically COM1 or COM2). See Figure 3-5.

3. Connect a network cable to the **Ethernet** connector on the back of the MultiVOIP. Connect the other end of the cable to your network.

4. If you intend to configure the MultiVOIP remotely using the MultiVOIP Windows GUI, connect an RJ-11 phone cable between the Command Modem connector (at the rear of the MultiVOIP) and a receptacle served by a telco POTS line. See Figure 3-6.

   The Command Modem is built into the MultiVOIP unit. To configure the MultiVOIP remotely using its Windows GUI, you must call into the MultiVOIP's Command Modem. Once a connection is made, the configuration process is identical to local configuration with the Windows GUI.

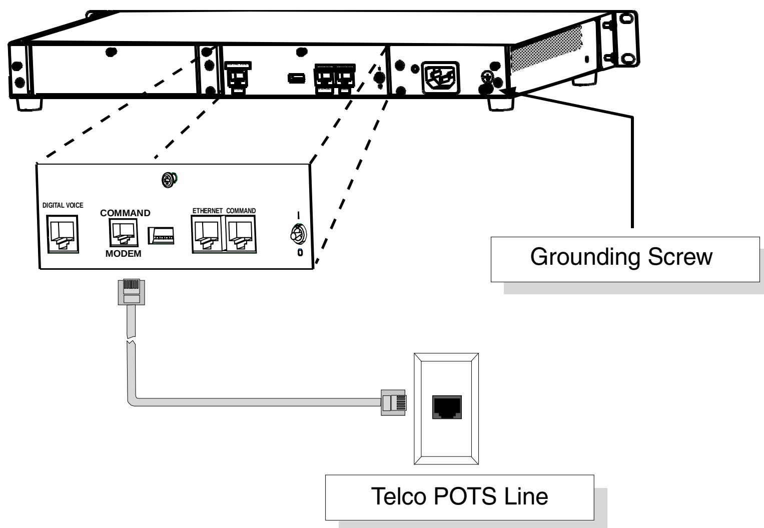


Figure 3-6. MVP-2410/3010 Voip Connections for GND & Remote Config Modem

5. Ensure that the unit is properly connected to earth ground by verifying that it is reliably grounded when mounted within a rack.

   This can be accomplished by connecting a grounding wire between the chassis grounding screw (see Figure 3-6) and a metallic object that will provide an electrical ground.

6. Turn on power to the MultiVOIP by setting the power switch on the right side panel to the **ON** position. Wait for the **Boot** LED on the MultiVOIP to go off before proceeding. This may take a couple of minutes.

Proceed to Chapter 4 "Software Installation."

Chapter 4: Software Installation

Introduction

Configuring software for your MultiVOIP entails three tasks:
(1) loading the software onto the PC (this is "Software Installation and is discussed in this chapter),
(2) setting values for telephony and IP parameters that will fit your system (this is "Technical Configuration" and it is discussed in Chapter 5), and
(3) establishing "phonebooks" that contain the various dialing patterns for VOIP calls made to different locations (this is "Phonebook Configuration" and it is discussed in Chapter 6 for North American (T1) telephony standards and in Chapter 7 for European (E1) telephony standards.

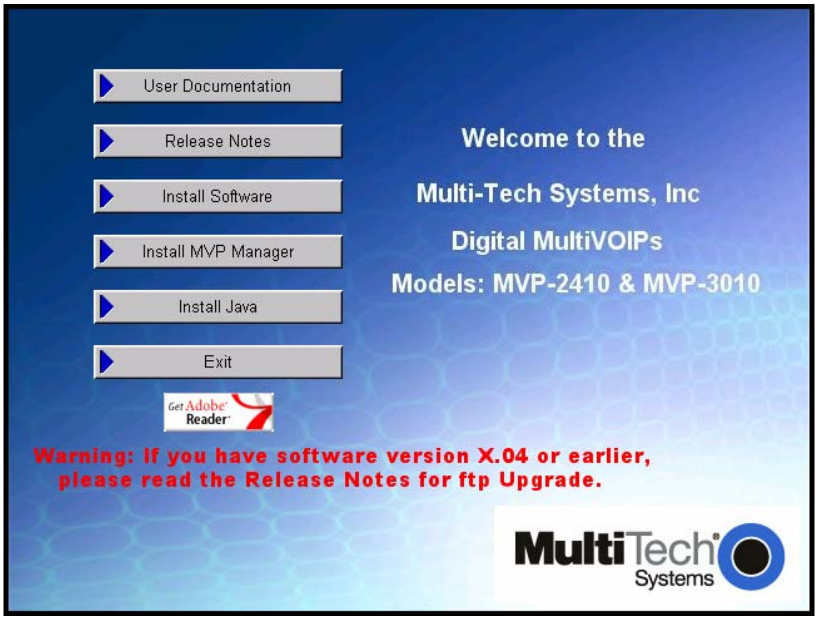
Loading MultiVOIP Software onto the PC

The software loading procedure does not present every screen or option in the loading process. It is assumed that someone with a thorough knowledge of Windows and the software loading process is performing the installation.

The MultiVOIP software and User Guide are contained on the MultiVOIP product CD. Because the CD is auto-detectable, it will start up automatically when you insert it into your CD-ROM drive. When you have finished loading your MultiVOIP software, you can view and print the User Guide by clicking on the **View Manuals** icon.

1. Be sure that your MultiVOIP has been properly cabled and that the power is turned on.

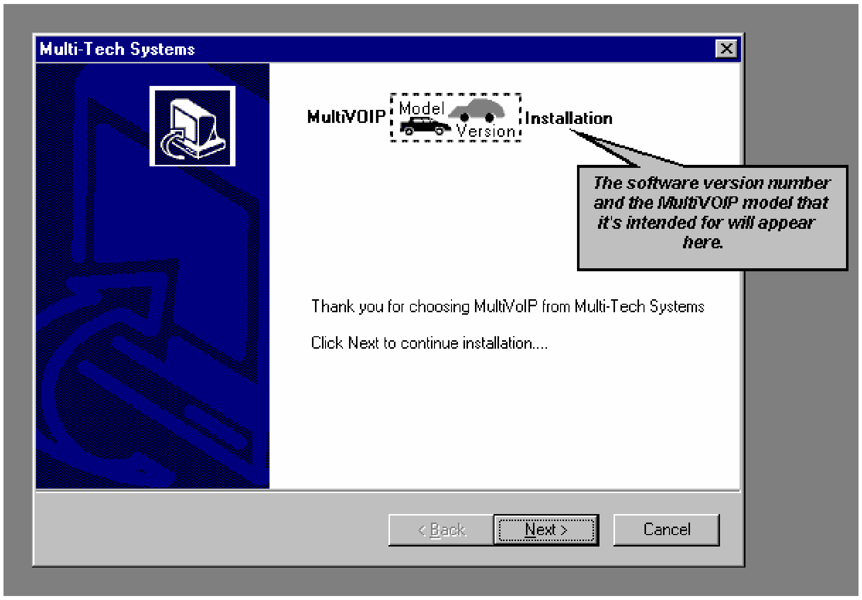
2. Insert the MultiVOIP CD into your CD-ROM drive. The CD should start automatically. It may take 10 to 20 seconds for the Multi-Tech CD installation window to display.



If the Multi-Tech Installation CD window does not display automatically, click **My Computer**, then right click the **CD ROM drive** icon, click **Open**, and then click the **Autorun** icon.

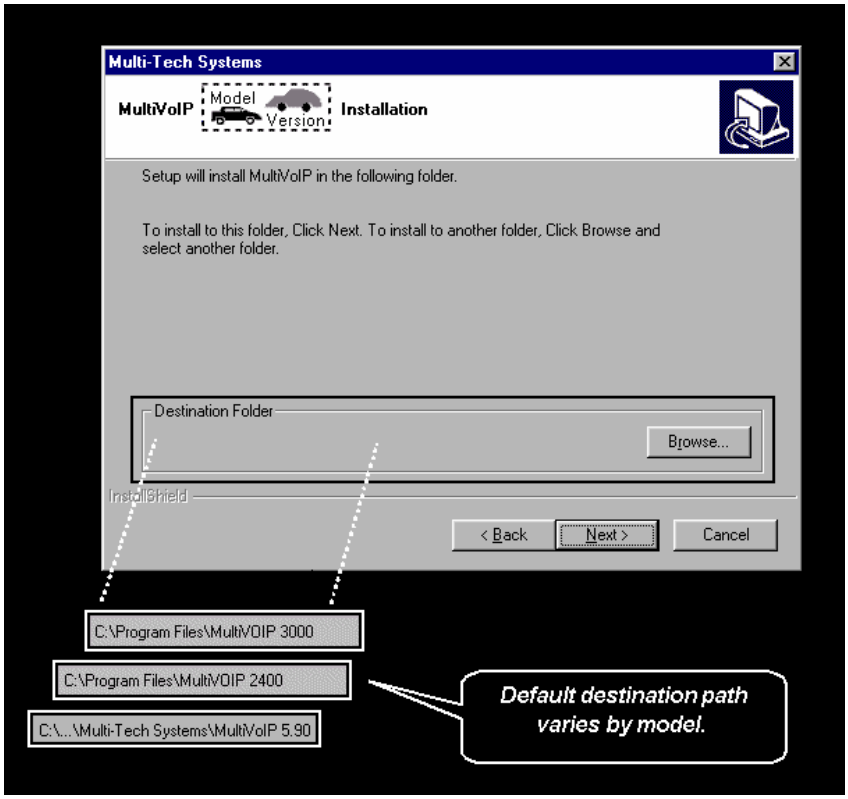
3. When the Multi-Tech Installation CD dialog box appears, click the **Install Software** icon.

4. A 'welcome' screen appears.



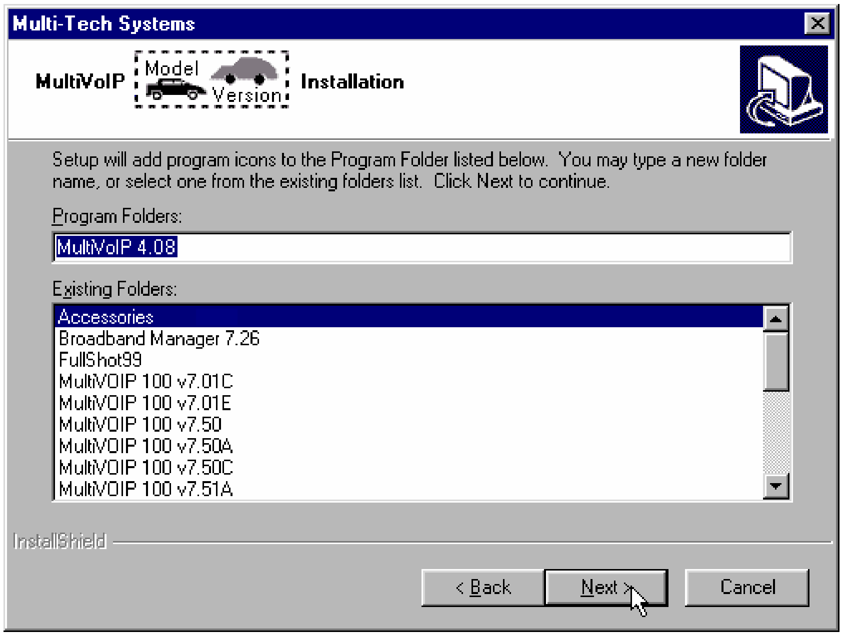
Press **Enter** or click **Next** to continue.

5. Follow the on-screen instructions to install your MultiVOIP software. The first screen asks you to choose the folder location of the files of the MultiVOIP software.



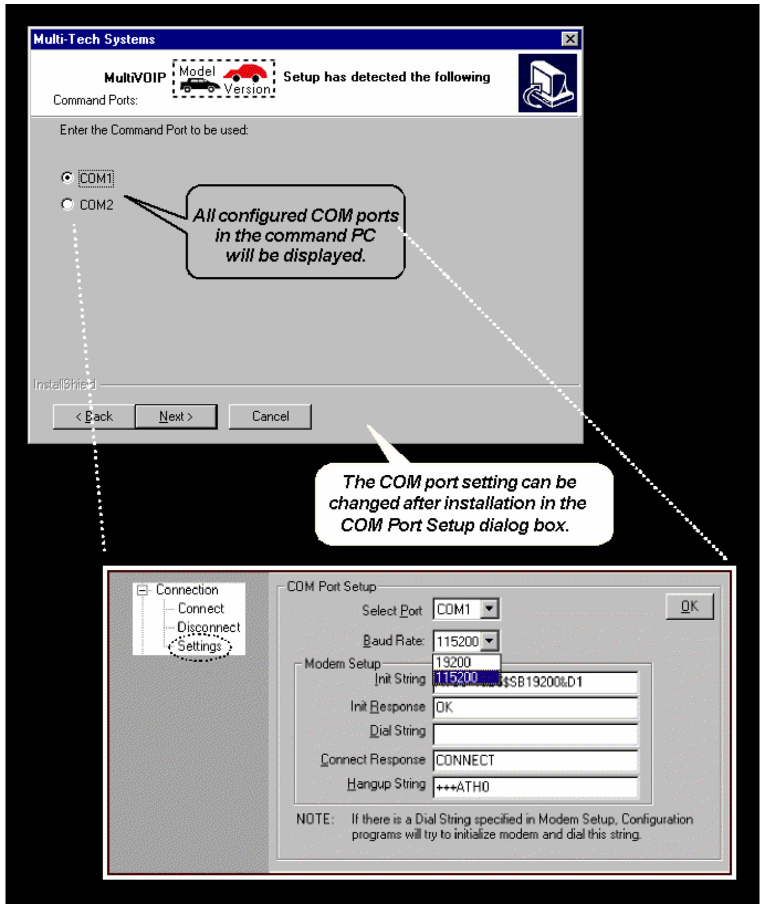
Choose a location and click **Next.**

6. At the next screen, you must select a program folder location for the MultiVOIP software program icon.



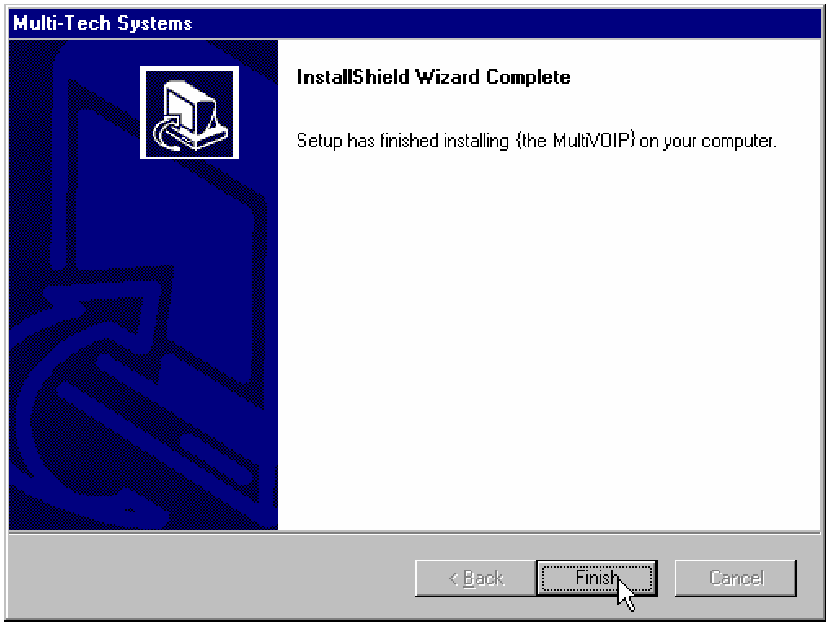
Click **Next**. Transient progress screens will appear while files are being copied.

7. On the next screen you can select the COM port that the command PC will use when communicating with the MultiVoip unit. After software installation, the COM port can be re-set in the MultiVOIP Software (from the sidebar menu, select **Connection | Settings** to access the **COM Port Setup** screen or use the keyboard shortcut Ctrl + G).



NOTE: If the COM port setting made here conflicts with the actual COM port resources available in the command PC, this error message will appear when the MultiVOIP program is launched. If this occurs, you must reset the COM port.	

8. A completion screen will appear.



Click **Finish**.

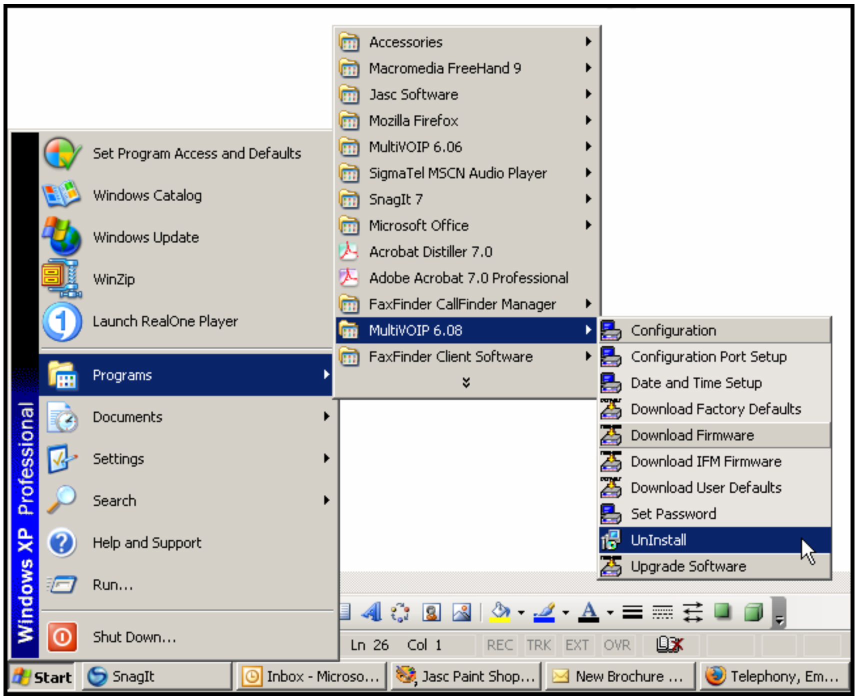
9. When setup of the MultiVOIP software is complete, you will be prompted to run the MultiVOIP software to configure the VOIP.

Software installation is complete at this point. You may proceed with Technical Configuration now or not, at your convenience.

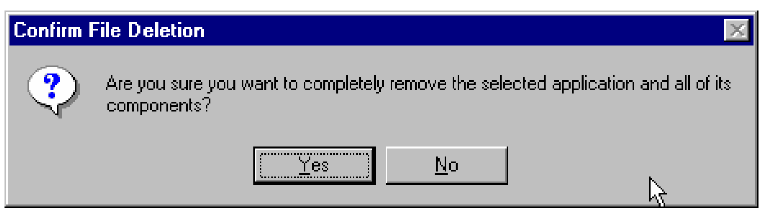
Technical Configuration instructions are in the next chapter of this manual.

Un-Installing the MultiVOIP Configuration Software

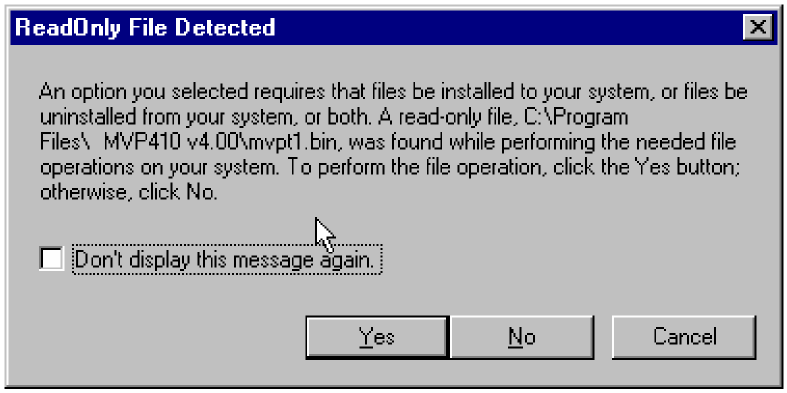
1. To un-install the MultiVOIP configuration software, go to **Start** | **Programs** and locate the entry for the MultiVOIP program. Select **Uninstall**.



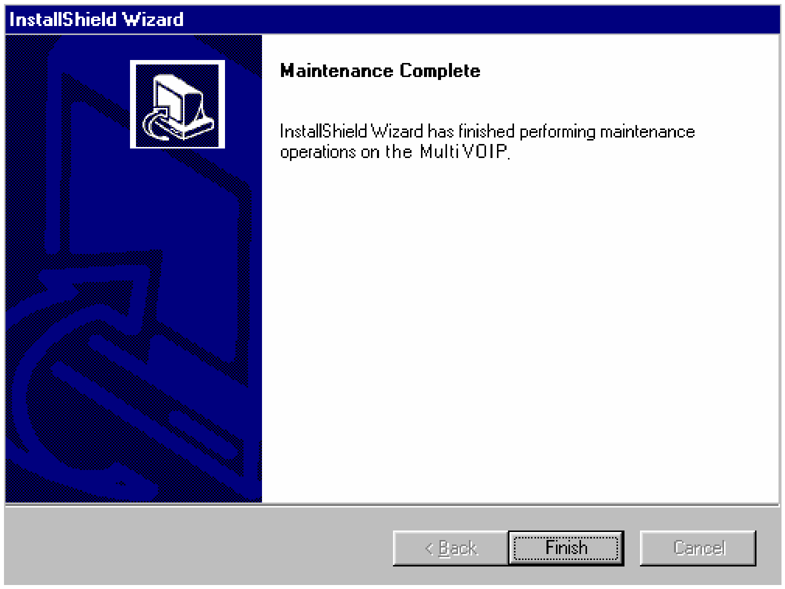
2. Two confirmation screens will appear. Click **Yes** and **OK** when you are certain you want to continue with the uninstallation process.



3. A special warning message similar to that shown below may appear concerning the MultiVOIP software's ".bin" file. Click **Yes**.



4. A completion screen will appear.



Click **Finish**.

Chapter 5: Technical Configuration

Configuring the MultiVOIP

There are two ways in which the MultiVOIP must be configured before operation: technical configuration and phonebook configuration.

Technical Configuration. First, the MultiVOIP must be configured to operate with technical parameter settings that will match the equipment with which it interfaces. There are eight types of technical parameters that must be set.

These technical parameters pertain to
(1) its operation in an IP network,
(2) its operation with telephony equipment,
(3) its transmission of voice and fax messages,
(4) its interaction with SNMP (Simple Network Management Protocol) network management software (MultiVoipManager),
(5) certain telephony attributes that are common to particular nations or regions,
(6) its operation with a mail server on the same IP network (per SMTP parameters) such that log reports about VoIP telephone call traffic can be sent to the administrator by email,
(7) implementing some common premium telephony features (Call Transfer, Call Hold, Call Waiting, Call ID – "Supplementary Services"), and
(8) selecting the method by which log reports will be made accessible.

The process of specifying values for the various parameters in these seven categories is what we call "technical configuration" and it is described in this chapter.

Phonebook Configuration. The second type of configuration that is required for the MultiVOIP pertains to the phone number dialing sequences that it will receive and transmit when handling calls. Dialing patterns will be affected by both the PBX/telephony equipment and the other VOIP devices that the MultiVOIP unit interacts with. We call this "Phonebook Configuration," and, for analog MultiVOIP units, it is described in Chapter 6. The *Quick Start Guide* presents additional information on phonebook setup.

Local/Remote Configuration. The MultiVOIP must be configured locally at first (to establish an IP address for the MultiVOIP unit). But changes to this initial configuration can be done either locally or remotely.

Local configuration is done through a connection between the "Command" port of the MultiVOIP and the COM port of the computer; the MultiVOIP configuration program is used.

Remote configuration is done through a connection between the MultiVOIP's Ethernet (network) port and a computer connected to the same network. The computer could be miles or continents away from the MultiVOIP itself. There are two ways of doing remote configuration and operation of the MultiVOIP unit: (1) using the MultiVoipManager SNMP program, or (2) using the MultiVOIP web browser interface program.

MultiVoipManager. MultiVoipManager is an SNMP agent program (Simple Network Management Protocol) that extends the capabilities of the MultiVOIP configuration program: MultiVoipManager allows the user to manage any number of VOIPs on a network, whereas the MultiVOIP configuration program can manage only the VOIP to which it is directly/locally connected. The MultiVoipManager can configure multiple VOIPs simultaneously, whereas the MultiVOIP configuration program can configure only one at a time.

MultiVoipManager may (but does not need to) reside on the same PC as the MultiVOIP configuration program. The MultiVoipManager program is on the MultiVOIP Product CD. Updates, when applicable, may be posted at on the MultiTech FTP site. To download, go to ftp://ftp.multitech.com/MultiVoip/.

Web Browser Interface. The MultiVOIP web browser GUI gives access to the same commands and configuration parameters as are available in the MultiVOIP Windows GUI except for logging functions. When using the web browser GUI, logging can be done by email (the SMTP option).

Functional Equivalence of Interfaces. The MultiVOIP configuration program is required to do the initial configuration (that is, setting an IP address for the MultiVOIP unit) so that the VOIP unit can communicate with the MultiVoipManager program or with the web browser GUI. Management of the VOIP after that point can be done from any of these three programs since they all offer essentially the same functionality. Functionally, either the MultiVoipManager program or the web browser GUI can replace the MultiVOIP configuration program after the initial configuration is complete (with minor exceptions, as noted).

WARNING: Do not attempt to interface the MultiVOIP unit with two control programs simultaneously (that is, by accessing the MultiVOIP configuration program via the Command Port and either the MultiVoipManager program or the web browser interface via the Ethernet Port). The results of using two programs to control a single VOIP simultaneously would be unpredictable.

Local Configuration

This manual primarily describes local configuration with the Windows GUI. After IP addresses have been set locally using the Windows GUI, most aspects of configuration (logging functions are an exception) can be handled through the web browser GUI, as well (see the *Operation and Maintenance* chapter of this manual). In most aspects of configuration, the Windows GUI and web-browser GUI differ only graphically, not functionally. For information on SNMP remote configuration and management, see the MultiVoipManager documentation.

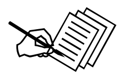
Pre-Requisites

To complete the configuration of the MultiVOIP unit, you ***must*** know several things about the overall system.

Before configuring your MultiVOIP Gateway unit, you must know the values for several IP and telephone parameters that describe the IP network system and telephony system (PBX or telco central office equipment) with which the digital MultiVOIP will interact. If you plan to receive log reports on phone traffic by email (SMTP), you must arrange to have an email address assigned to the VOIP unit on the email server on your IP network. A summary of this configuration information appears on page 58 (“Config Info CheckList”).

IP Parameters

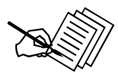
The following parameters must be known about the network (LAN, WAN, Internet, etc.) to which the MultiVOIP will connect:

➸	*Ask your computer network administrator.*	*Info needed to operate:* **all MultiVOIP models.**
	 IP Network Parameters: Record for each VOIP Site in System	
	• IP Address	
	• IP Mask	
	• Gateway	
	• Domain Name Server (DNS) Info	
	• If SIP protocol is used, determine whether or not 802.1p Packet Prioritization will be used.	

Write down the values for these IP parameters. You will need to enter these values in the "IP Parameters" screen in the Configuration section of the MultiVOIP software. You must have this IP information about *every* VOIP in the system.

T1 Telephony Parameters (for MVP2410)

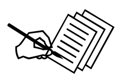
The following parameters must be known about the PBX or telco central office equipment to which the T1 MultiVOIP will connect:

➸	**T1 Phone Parameters** *Ask phone company or PBX maintainer.*	*Info needed to operate:* **MVP2410**
	 T1 Telephony Parameters: Record for this VOIP Site	
	• Which frame format is used? ESF___ or D4___	
	• Which CAS or PRI protocol is used? _______________	
	• Clocking: Does the PBX or telco switch use internal or external clocking? __________________ Note that the setting used in the voip unit will be the opposite of the setting used by the telco/PBX.	
	• Which line coding is used? AMI___ or B8ZS___	

Write down the values for these T1 parameters. You will need to enter these values in the "T1/E1 Parameters" screen in the Configuration section of the MultiVOIP software.

E1 Telephony Parameters (for MVP3010)

The following parameters must be known about the PBX or telco central office equipment to which the E1 MultiVOIP will connect:

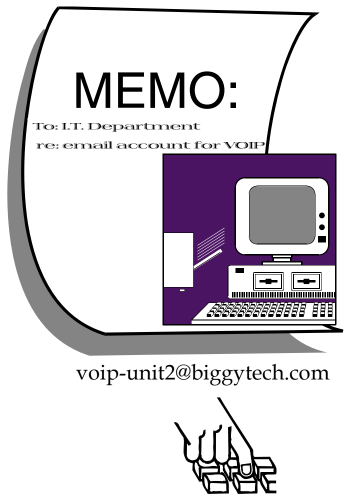
➸	**E1 Phone Parameters** *Ask phone company or PBX maintainer.*	*Info needed to operate:* **MVP3010**
	 E1 Telephony Parameters: Record for this VOIP Site	
	• Which frame format is used? Double Frame_____ MultiFrame w/ CRC4_____ MultiFrame w/ CRC4 modified_____	
	• Which CAS or PRI protocol is used? _______________	
	• Clocking: Does the PBX or telco switch use internal or external clocking? __________________ Note that the setting used in the voip unit will be the opposite of the setting used by the telco/PBX.	
	• Which line coding is used? AMI___ or HDB3___	
	• Pulse shape level?: (most commonly 0 to 40 meters)	

Write down the values for these E1 parameters. You will need to enter these values in the "T1/E1 Parameters" screen in the Configuration section of the MultiVOIP software.

SMTP Parameters (for email call log reporting)

<table>
<tr><td>required if log reports of VOIP call traffic are to be sent by email</td><td>Optional</td></tr>
<tr><td>SMTP Parameters Preparation Task:

Ask Mail Server administrator to set up email account (with password) for the MultiVOIP unit itself. Be sure to give a unique identifier to each individual MultiVOIP unit. .

Get the IP address of the mail server computer, as well.</td><td>
</td></tr>
</table>

Config Info CheckList

Type of Config Info Gathered √	Models to which Config Info applies	MultiVOIP Configuration screen on which to enter Config Info	Info Obtained	Info Entered
IP info for voip unit ● IP address ● Gateway ● DNS IP (if used) ● 802.1p Prioritization (if used)	MVP2410, MVP3010	Ethernet/IP Parameters		
Frame Format (Choices: ESF, D4, F4, SLC96)	MVP2410	T1/E1/ISDN Parameters		
Frame Format (Choices: Double Frame, Multi-Frame w/ CRC4, Multi-Frame w/ CRC4 Modified)	MVP3010	T1/E1/ISDN Parameters		
CAS Protocol (Choices: FXS Loop Start, E&M Wink, E&M Wink w/ Dial Tone, FXO Ground Start, FXO Loop Start, FXS Ground Start, E&M Immediate, MFR2-China, Clear Channel)	MVP2410, MVP3010	T1/E1/ISDN Parameters		
ISDN-PRI Protocol (only if ISDN-PRI is used) (Choices: Network, Terminal)	MVP2410, MVP3010	T1/E1/ISDN Parameters		
Clocking (Choices: Internal, External)	MVP2410 MVP3010	T1/E1/ISDN Parameters		
Line Coding (Choices: AMI, B8ZS)	MVP2410, MVP3010	T1/E1/ISDN Parameters		
Pulse Shape Level (Choices: 0 – 40 m)	MVP2410, MVP3010	T1/E1/ISDN Parameters		
Country Code	MVP2410 MVP3010	Regional Parameters		
Email address for voip (optional)	all	SMTP Parameters		
Reminder: *Be sure to* ***Save Setup*** *after entering configuration values.*				

Local Configuration Procedure (Summary)

After the MultiVOIP configuration software has been installed in the 'Command' PC (which is connected to the MultiVOIP unit), several steps must be taken to configure the MultiVOIP to function in its specific setting. Although the summary below includes all of these steps, some are optional.

1. Check Power and Cabling.

2. Start MultiVOIP Configuration Program.

3. Confirm Connection.

4. Solve Common Connection Problems.

   A. Fixing a COM Port Problem.

   B. Fixing a Cabling Problem.

5. Familiarize yourself with configuration parameter screens and how to access them.

6. Set Ethernet/IP Parameters.

7. Set up web browser GUI (optional).

8. Set Voice/Fax Parameters.

9. Set T1/E1 Parameters.

10. Set ISDN Parameters (if applicable).

11. Set Call Signaling parameters. The choice of H.323, SIP, or SPP is made in the Outbound Phonebook, but details are configured in the Call Signaling Parameters screen.

12. Set SNMP Parameters (applicable if MultiVoipManager remote management software is used).

13. Set Regional Parameters (Phone Signaling Tones & Cadences and setup for built-in Remote Configuration/Command Modem).

13. Set Custom Tones and Cadences (optional).

14. Set SMTP Parameters (applicable if Log Reports are via Email).

15. Set Log Reporting Method (GUI, locally in MultiVOIP Configuration program; SNMP, remotely in MultiVoipManager program; or SMTP, via email).

16. Set Supplementary Services Parameters. The Supplementary Services screen allows voip deployment of features that are normally found in PBX or PSTN systems (e.g., call transfer and call waiting).

17. Set NAT Traversal (STUN) parameters. Optional. Applicable only under SIP Call Signaling when the UDP transport protocol is used.

18. Set RADIUS parameters. Optional. Used only if system interfaces with RADIUS server for billing or other accounting functions.

19. Set Baud Rate (of COM port connection to 'Command' PC).

20. View System Info screen and set updating interval (optional).

21. Save the MultiVOIP configuration.

22. Create a User Default Configuration (optional).

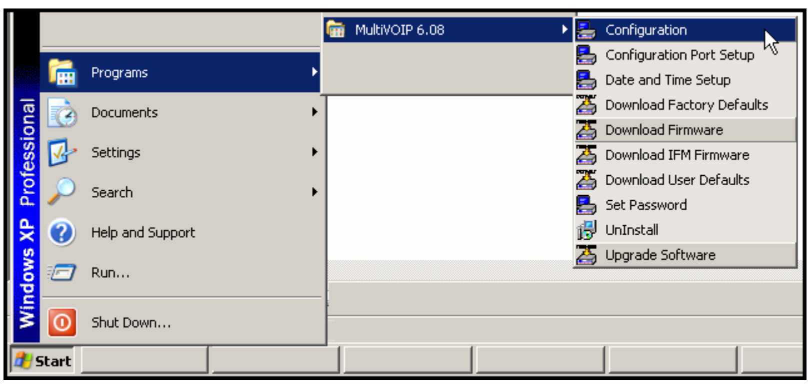
When technical configuration is complete, you will need to configure the MultiVOIP's inbound and outbound phonebooks. This manual has separate chapters describing ***T1 Phonebook Configuration*** for North-American-influenced telephony settings and ***E1 Phonebook Configuration*** for Euro-influenced telephony settings.

Local Configuration Procedure (Detailed)

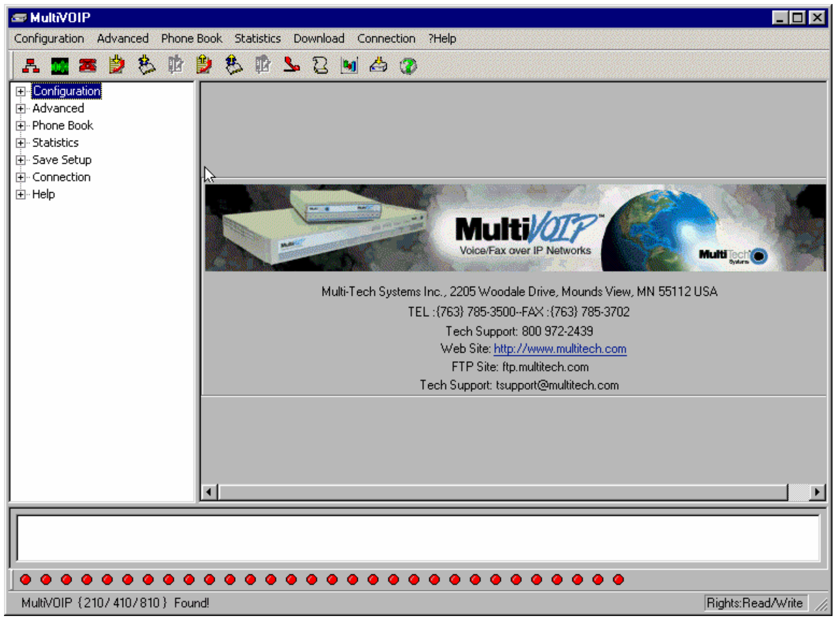
You can begin the configuration process as a continuation of the MultiVOIP software installation. You can establish your configuration or modify it at any time by launching the MultiVOIP program from the Windows **Start** menu.

1. **Check Power and Cabling**. Be sure the MultiVOIP is turned on and connected to the computer via the MultiVOIP's Command Port (DB9 connector at computer's COM port; RJ45 connector at MultiVOIP).

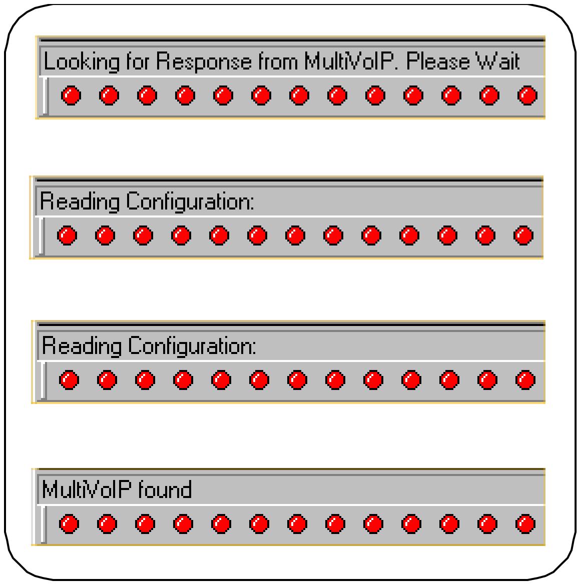
2. **Start MultiVOIP Configuration Program**. Launch the MultiVOIP program from the Windows **Start** menu (from the folder location determined during installation).



3. **Confirm Connection**. If the MultiVOIP is set for an available COM port and is correctly cabled to the PC, the MultiVOIP main screen will appear. (If the main screen appears *grayed out* and seems inaccessible, go to step 4.)



In the lower left corner of the screen, the connection status of the MultiVOIP will be displayed. The messages in the lower left corner will change as detection occurs. The message "MultiVOIP Found" confirms that the MultiVOIP is in contact with the MultiVOIP configuration program. Skip to step 5.

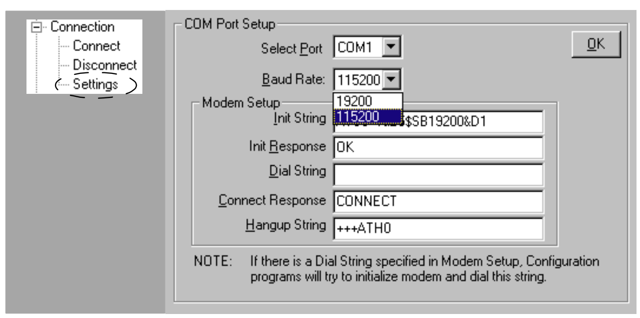


4. **Solving Common Connection Problems**.

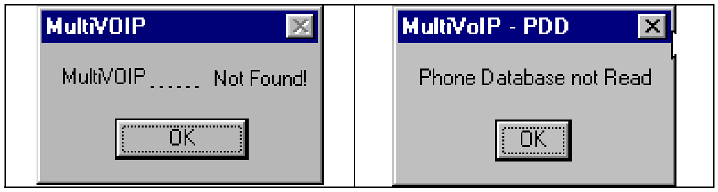
A. Fixing a COM Port Problem. If the MultiVOIP main screen appears but is grayed out and seems inaccessible, the COM port that was specified for its communication with the PC is unavailable and must be changed. An error message will appear.

To change the COM port setting, use the **COM Port Setup** dialog box, which is accessible via the keyboard shortcut **Ctrl + G** or by going to the **Connection** pull-down menu and choosing "Settings." In the "Select Port" field, select a COM port that is available on the PC. (If no COM ports are currently available, re-allocate COM port resources in the computer's MS Windows operating system to make one available.)

Ctrl + G



4B. Fixing a Cabling Problem. If the MultiVOIP cannot be located by the computer, two error messages will appear (saying "Multi-VOIP Not Found" and "Phone Database Not Read").

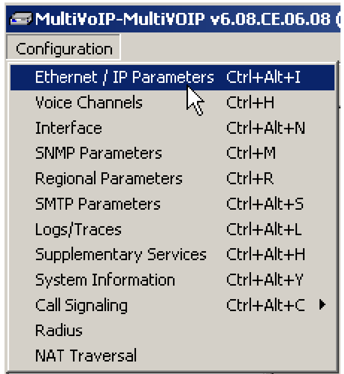
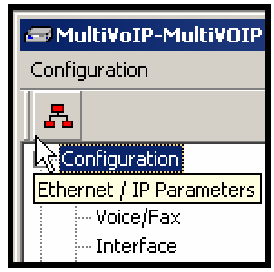
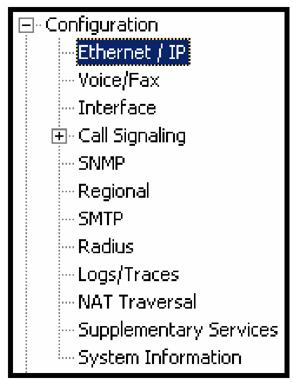


In this case, the MultiVOIP is simply disconnected from the network. For instructions on MultiVOIP cable connections, see the Cabling section of Chapter 3.

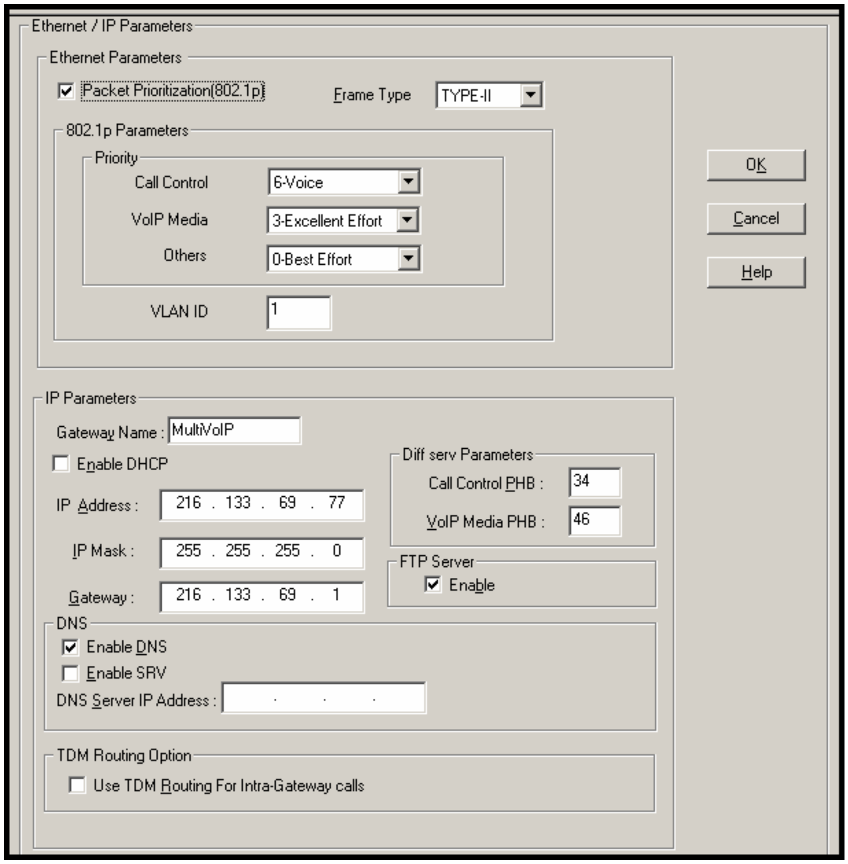
5. **Configuration Parameter Groups: Getting Familiar, Learning About Access**. The first part of configuration concerns IP parameters, Voice/FAX parameters, Telephony Interface parameters, SNMP parameters, Regional parameters, SMTP parameters, Supplementary Services parameters, Logs, and System Information. In the MultiVOIP software, these seven types of parameters are grouped together under "Configuration" and each has its own dialog box for entering values.

Generally, you can reach the dialog box for these parameter groups in one of four ways: pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

6. **Set Ethernet/IP Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing "Ethernet/IP Parameters"</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td>
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + Alt + I</td><td>
</td></tr>
</table>

In each field, enter the values that fit your particular network.



The **Ethernet/IP Parameters** fields are described in the tables and text passages below. Note that both DiffServ parameters (Call Control PHB and VoIP Media PHB) must be set to zero if you enable Packet Prioritization (802.1p). Nonzero DiffServ values negate the prioritization scheme.

Ethernet/IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Ethernet Parameters		
Packet Prioritization (802.1p)	Y/N	Select to activate prioritization under 802.1p protocol (described below). .
Frame Type	Type II, SNAP	Must be set to match network's frame type. Default is Type II.
802.1p	A draft standard of the IEEE about data traffic prioritization on Ethernet networks. The 802.1p draft is an extension of the 802.1D bridging standard. 802.1D determines how prioritization will operate within a MAC-layer bridge for any kind of media. The 802.1Q draft for virtual local-area-networks (VLANs) addresses the issue of prioritization for Ethernet networks in particular. 802.1p enacts this Quality-of-Service feature using 3 bits. This 3-bit code allows data switches to reorder packets based on priority level. The descriptors for the 8 priority levels are given below. **802.1p PRIORITY LEVELS** *LOWEST PRIORITY* 1 – **Background**: Bulk transfers and other activities permitted on the network, but should not affect the use of network by other users and applications. 2 – **Spare**: An unused (spare) value of the user priority. 0 – **Best Effort** (default): Normal priority for ordinary LAN traffic. 3 – **Excellent Effort**: The best effort type of service that an information services organization would deliver to its most important customers.	

Ethernet/IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Ethernet Parameters		
802.1p (continued)	4 – **Controlled Load**: Important business applications subject to some form of "Admission Control", such as preplanning of Network requirement, characterized by bandwidth reservation per flow. 5 – **Video**: Traffic characterized by delay < 100 ms. 6 – **Voice**: Traffic characterized by delay < 10 ms. 7 - **Network Control**: Traffic urgently needed to maintain and support network infrastructure. *HIGHEST PRIORITY*	
Call Control Priority	0-7, where 0 is lowest priority	Sets the priority for signaling packets.
VoIP Media Priority	0-7, where 0 is lowest priority	Sets the priority for media packets.
Others (Priorities)	0-7, where 0 is lowest priority	Sets the priority for SMTP, DNS, DHCP, and other packet types.
VLAN ID	1 - 4094	The 802.1Q IEEE standard allows virtual LANs to be defined within a network. This field identifies each virtual LAN by number.

Ethernet/IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
IP Parameter fields		
Gateway Name	alphanumeric	Descriptor of current voip unit to distinguish it from other units in system.
Enable DHCP	Y/N disabled by default	Dynamic Host Configuration Protocol is a method for assigning IP address and other IP parameters to computers on the IP network in a single message with great flexibility. IP addresses can be static or temporary depending on the needs of the computer.
IP Address	4-places, 0-255	The unique LAN IP address assigned to the MultiVOIP.
IP Mask	4-places, 0-255	Subnetwork address that allows for sharing of IP addresses within a LAN.
Gateway	4-places, 0-255.	The IP address of the device that connects your MultiVOIP to the Internet.

Ethernet/IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
DiffServ Parameter fields	DiffServ PHB (Per Hop Behavior) values pertain to a differential prioritizing system for IP packets as handled by DiffServ-compatible routers. There are 64 values, each with an elaborate technical description. These descriptions are found in TCP/IP standards RFC2474, RFC2597, and, for present purposes, in RFC3246, which describes the value 34 (34 decimal; 22 hex) for Assured Forwarding behavior (default for Call Control PHB) and the value 46 (46 decimal; 2E hexadecimal) for Expedited Forwarding behavior (default for Voip Media PHB). Before using values other than these default values of 34 and 46, consult these standards documents and/or a qualified IP telecommunications engineer. To disable DiffServ, configure both fields to 0 decimal. The next page explains DiffServ in the context of the IP datagram.	
Call Control PHB	0 – 63 default = 34 .	Value is used to prioritize call setup IP packets.
Voip Media PHB	0 – 63 default = 46 *n*	Value is used to prioritize the RTP/RTCP audio IP packets.

The IP Datagram with Header, Its Type-of-Service field, & DiffServ

bits =>

<table>
<tr><td>0</td><td>4</td><td>8</td><td>16</td><td>19</td><td>24</td><td>31</td></tr>
<tr><td>VERS</td><td>HLEN</td><td>TYPE OF SERVICE</td><td colspan="4">TOTAL LENGTH</td></tr>
<tr><td colspan="3">IDENTIFICATION</td><td>FLAGS</td><td colspan="3">FRAGMENT OFFSET</td></tr>
<tr><td colspan="2">TIME TO LIVE</td><td>PROTOCOL</td><td colspan="4">HEADER CHECKSUM</td></tr>
<tr><td colspan="7">SOURCE IP ADDRESS</td></tr>
<tr><td colspan="7">DESTINATION IP ADDRESS</td></tr>
<tr><td colspan="5">IP OPTIONS (if any)</td><td colspan="2">PADDING …
end of header</td></tr>
<tr><td colspan="7">DATA</td></tr>
<tr><td colspan="7">…</td></tr>
</table>

The TOS field consists of eight bits, of which only the first six are used. These six bits are called the “Differentiated Service Codepoint” or DSCP bits.

The Type of Service or “TOS” field

<table>
<tr><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td></tr>
<tr><td colspan="3">PRECEDENCE</td><td>D</td><td>T</td><td>R</td><td colspan="2">unused</td></tr>
</table>

three precedence have eight values, 0-7, ranging from “normal” precedence (value of 0) to “network control” (value of 7). When set, the *D* bit requests low delay, the *T* bit requests high throughput, and the *R* bit requests high reliability.

Routers that support DiffServ can examine the six DSCP bits and prioritize the packet based on the DSCP value. The DiffServ Parameters fields in the MultiVOIP IP Parameters screen allow you to configure the DSCP bits to values supported by the router. Specifically, the Voip Media PHB field relates to the prioritizing of audio packets (RTP and RTCP packets) and the Call Control PHB field relates to the prioritzing of non-audio packets (packets concerning call set-up and tear-down, gatekeeper registration, etc.).

The MultiVOIP Call Control PHB parameter defaults to 34 decimal (22 hex; 100010 binary – consider vis-à-vis TOS field above) for Assured Forwarding behavior. The MultiVOIP Voip Media PHB parameter defaults to the value 46 decimal (2E hex; 101110 binary – consider vis-à-vis TOS field above). To disable DiffServ, configure both fields to 0 decimal.

Ethernet/IP Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
FTP Parameter fields		
FTP Server Enable	Y/N Default = disabled See "FTP Server File Transfers" in *Operation & Maintenance* chapter.	MultiVOIP unit has an FTP Server function so that firmware and other important operating software files can be transferred to the voip via the network.
DNS Parameter fields		
Enable DNS	Y/N Default = disabled	Enables Domain Name Space/System function where computer names are resolved using a worldwide distributed database.
Enable SRV	Y/N	Enables 'service record' function. Service record is a category of data in the Internet Domain Name System specifying information on available servers for a specific protocol and domain, as defined in RFC 2782. Newer internet protocols like SIP, STUN, H.323, POP3, and XMPP may require SRV support from clients. Client implementations of older protocols, like LDAP and SMTP, may have been enhanced in some settings to support SRV.
DNS Server IP Address	4-places, 0-255.	IP address of specific DNS server to be used to resolve Internet computer names.

About Service Records

An SRV record holds the following information:

- **Service**: the symbolic name of the desired service.
- **Protocol**: this is usually either TCP or UDP.
- **Domain name**: the domain for which this record is valid.
- **TTL**: standard DNS time to live field.
- **Class**: standard DNS class field (this is always *IN*).
- **Priority**: the priority of the target host.
- **Weight**: A relative weight for records with the same priority.
- **Port**: the TCP or UDP port on which the service is to be found.
- **Target**: the hostname of the machine providing the service.

An example SRV record might look like this:

_sip._tcp.example.com 86400 IN SRV 0 5 5060 sipserver.example.com.

This expression denotes a server named sipserver.example.com. This server listens on TCP port 5060 for SIP protocol connections. The priority given here is 0, and the weight is 5.

TDM Routing Option Parameter fields		
Use TDM Routing for Intra-Gateway calls	Y/N; enabled by default	Allows calls placed between ports on the same MultiVOIP voice channel board to be routed over internal Time Division Multiplex bus without conversion to IP. TDM routing effectively eliminates the delay introduced by IP conversion. If you require all calls to be IP routed, disable the “use TDM Routing for Intra-Gateway Calls” option. Since this is not normally required, we generally recommend leaving TDM Routing enabled.

7. **Set up the Web Browser GUI (Optional)**. After an IP address for the MultiVOIP unit has been established, you can choose to do any further configuration of the unit (a) by using the MultiVOIP web browser GUI, or (b) by continuing to use the MultiVOIP Windows GUI. If you want to do configuration work using the web browser GUI, you must first set it up. To do so, follow the steps below.

A. Set IP address of MultiVOIP unit using the MultiVOIP Configuration program (the Windows GUI).

B. Save Setup in Windows GUI.

C. Close Windows GUI.

D. Install Java program from MultiVOIP product CD (on first use only).

E. Open web browser.

F. Browse to IP address of MultiVOIP unit.

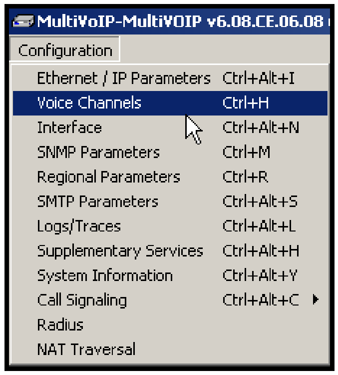
G. If username and password have been established, enter them when when prompted.

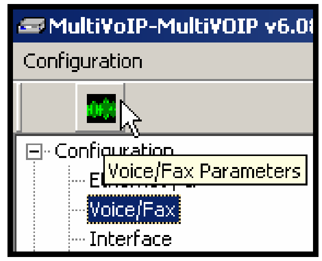
H. Set browser to allow pop-ups. The MultiVOIP Web GUI makes extensive use of pop-up windows to access screens and commands.

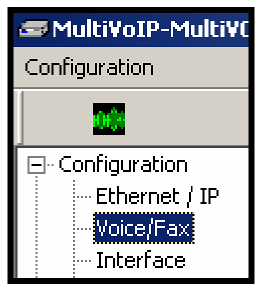
I. Use web browser GUI to configure or operate MultiVOIP unit. The configuration screens in the web browser GUI will have the same content as their counterparts in the Windows GUI; only the graphic presentation will be different.

For more details on enabling the MultiVOIP web GUI, see the "Web Browser Interface" section of the *Operation & Maintenance* chapter of this manual.

8. **Set Voice/FAX Parameters.** This dialog box can be reached by pulldown menu, toolbar icon, keyboard shortcut, or sidebar.

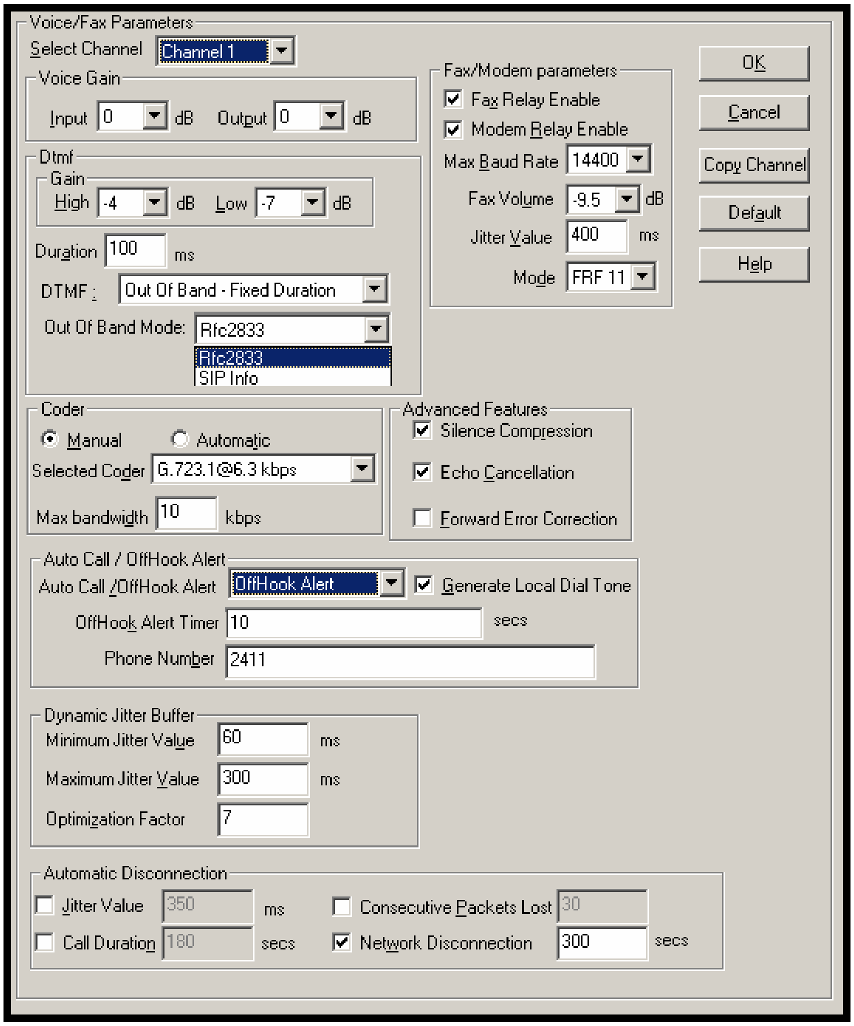
<table>
<tr><th colspan="2">Accessing “Voice/FAX Parameters”</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>


</td><td>


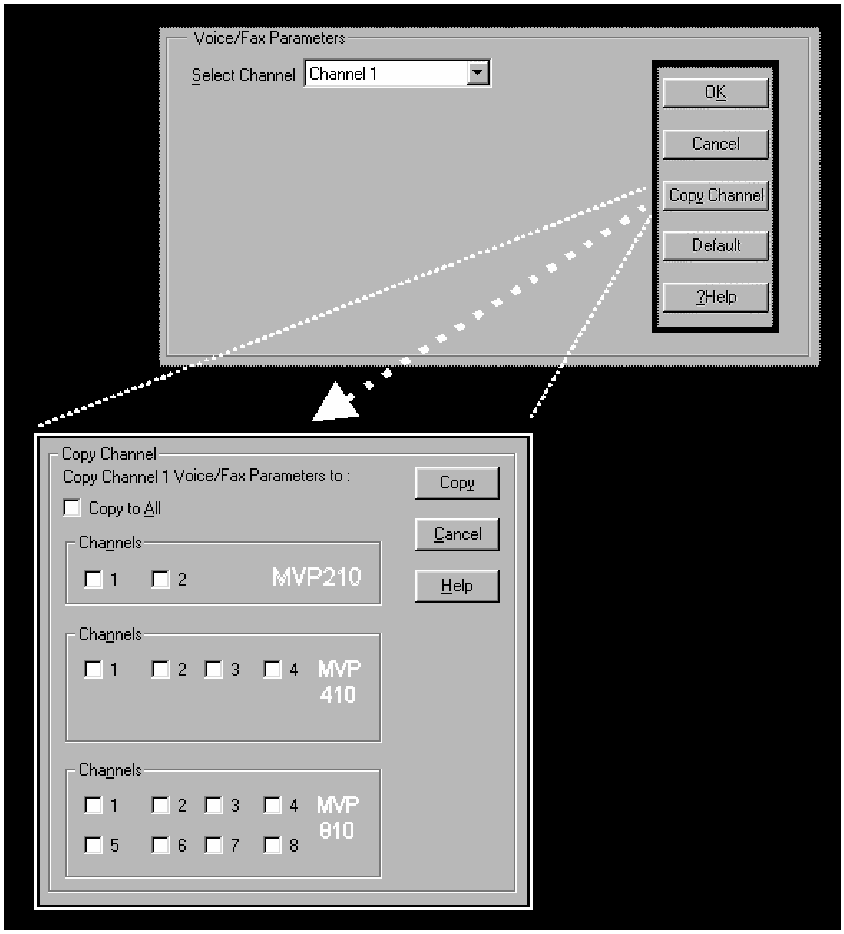
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + H</td><td>


</td></tr>
</table>

In each field, enter the values that fit your particular network.



Note that Voice/FAX parameters are applied on a channel-by-channel basis. However, once you have established a set of Voice/FAX parameters for a particular channel, you can apply this entire set of Voice/FAX parameters to another channel by using the **Copy Channel** button and its dialog box. To copy a set of Voice/FAX parameters to all channels, select "Copy to All" and click **Copy**.



The **Voice/FAX Parameters** fields are described in the tables below.

Voice/Fax Parameter Definitions		
Field Name	**Values**	**Description**
Default	--	When this button is clicked, all Voice/FAX parameters are set to their default values.
Select Channel	1-2 (210) 1-4 (410) 1-8 (810)	Channel to be configured is selected here.
Copy Channel	--	Copies the Voice/FAX attributes of one channel to another channel. Attributes can be copied to multiple channels or all channels at once.
Voice Gain	--	Signal amplification (or attenuation) in dB.
Input Gain	+31dB to –31dB	Modifies audio level entering voice channel before it is sent over the network to the remote VOIP. The default & recommended value is **0 dB**.
Output Gain	+31dB to –31dB	Modifies audio level being output to the device attached to the voice channel. The default and recommended value is **0 dB**.
DTMF Parameters		
DTMF Gain	--	The **DTMF Gain** (Dual Tone Multi-Frequency) controls the volume level of the DTMF tones sent out for Touch-Tone dialing.
DTMF Gain, High Tones	+3dB to -31dB & “mute”	Default value: **-4 dB**. Not to be changed except under supervision of MultiTech’s Technical Support.
DTMF Gain, Low Tones	+3dB to -31dB & “mute”	Default value: **-7 dB**. Not to be changed except under supervision of MultiTech’s Technical Support.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
DTMF Parameters		
Duration (DTMF)	60 – 3000 ms	When **DTMF: Out of Band** is selected, this setting determines how long each DTMF digit 'sounds' or is held. Default = 100 ms. Not supported in 5.02c BRI software.
DTMF In/Out of Band	Out of Band, or Inband	When **DTMF Out of Band** is selected, the MultiVOIP detects DTMF tones at its input and regenerates them at its output. When **DTMF Inband** is selected, the DTMF digits are passed through the MultiVOIP unit as they are received. In 502c BRI software, "DTMF Out of Band" can be checked or unchecked.
Out of Band Mode	RFC 2833, SIP Info	**RFC2833 method**. Uses an RTP mode defined in RFC 2833 to transmit the DTMF digits. **SIP Info method**. Generates dual tone multi frequency (DTMF) tones on the telephony call leg. The SIP INFO message is sent along the signaling path of the call. You must set this parameter per the capabilities of the remote endpoint with which the voip will communicate. The RFC2833 method is the more common of the two methods.
FAX Parameters		
Fax Enable	Y/N	Enables or disables fax capability for a particular channel.
Modem Relay Enable	Y/N	When enabled, modem traffic can be carried on voip system. When disabled, modem traffic will bypass the voip system (Modem Bypass mode).
Max Baud Rate (Fax)	2400, 4800, 7200, 9600, 12000, 14400 bps	Set to match baud rate of fax machine connected to channel (see Fax machine's user manual). Default = 14400 bps.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Valuee**	**Description**
FAX Parameters (cont'd)		
Fax Volume (Default = -9.5 dB)	-18.5 dB to –3.5 dB	Controls output level of fax tones. To be changed only under the direction of Multi-Tech's Technical Support.
Jitter Value (Fax)	Default = 400 ms	Defines the inter-arrival packet deviation (in milliseconds) for the fax transmission. A higher value will increase the delay, allowing a higher percentage of packets to be reassembled. A lower value will decrease the delay allowing fewer packets to be reassembled.
Mode (Fax)	FRF 11; T.38 (T.38 not currently sup-ported)	**FRF11** is frame-relay FAX standard using these coders: G.711, G.728, G.729, G.723.1. **T.38** is an ITU-T standard for storing and forwarding FAXes via email using X.25 packets. It uses T.30 fax standards and includes special provisions to preclude FAX timeouts during IP transmissions.

Voice/Fax Parameter Definitions (cont'd)		
Coder Parameters		
Coder	Manual or Auto-matic	Determines whether selection of coder is manual or automatic. When Automatic is selected, the local and remote voice channels will negotiate the voice coder to be used by selecting the highest bandwidth coder supported by both sides without exceeding the Max Bandwidth setting. G.723, G.729, or G.711 are negotiated.
Selected Coder	**G.711** a/u law 64 kbps; **G.726**, @ 16/24/32 /40 kbps; **G.727**, @ nine bps rates; **G.723.1** @ 5.3 kbps, 6.3 kbps; **G.729**, 8kbps; **Net Coder** @ 6.4, 7.2, 8, 8.8, 9.6 kbps	Select from a range of coders with specific bandwidths. The higher the bps rate, the more bandwidth is used. The channel that you are calling must have the same voice coder selected. Default = G.723.1 @ 6.3 kbps, as required for H.323. Here 64K of digital voice are compressed to 6.3K, allowing several simultaneous conversations over the same bandwidth that would otherwise carry only one. To make selections from the Selected Coder drop-down list, the Manual option must be enabled.
Max bandwidth (coder)	11 – 128 kbps	This drop-down list enables you to select the maximum bandwidth allowed for this channel. The Max Bandwidth drop-down list is enabled only if the Coder is set to Automatic. If coder is to be selected automatically ("Auto" setting), then enter a value for maximum bandwidth.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Advanced Features		
Silence Compression	Y/N	Determines whether silence compression is enabled (checked) for this voice channel. With **Silence Compression** enabled, the MultiVOIP will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel. Default = on.
Echo Cancellation	Y/N	Determines whether echo cancellation is enabled (checked) for this voice channel. **Echo Cancellation** removes echo and improves sound quality. Default = on.
Forward Error Correction	Y/N	Determines whether forward error correction is enabled (checked) for this voice channel. **Forward Error Correction** enables some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel. Default = Off

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
AutoCall/Offhook Alert Parameters		
Auto Call / Offhook Alert	AutoCall, Offhook Alert	The **AutoCall** option enables the local MultiVOIP to call a remote MultiVOIP without the user having to dial a Phone Directory Database number. As soon as you access the local MultiVOIP voice/fax channel, the MultiVOIP immediately connects to the remote MultiVOIP identified in the **Phone Number** box of this option. If the "Pass Through Enable" field is checked in the Interface Parameters screen, AutoCall must be used. The Offhook Alert option applies only to FXS channels. The **Offhook Alert** option works like this: if a phone goes offhook and yet no number is dialed within a specific period of time (as set in the **Offhook Alert Timer** field), then that phone will automatically dial the Alert phone number for the voip channel. (The Alert phone number must be set in the **Voice/Fax Parameters | Phone Number** field; if the voip system is working without a gatekeeper unit, there must also be a matching phone number entry in the Outbound Phonebook.). One use of this feature would be for emergency use where a user goes off hook but does not dial, possibly indicating a crisis situation. The Offhook Alert feature uses the **Intercept Tone**, as listed in the **Regional Parameters** screen. This tone will be outputted on the phone that was taken off hook but that did not dial. The other end of the connection will hear audio from the "crisis" end as is it would during a normal phone call.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
AutoCall/Offhook Alert Parameters		
Auto Call / Offhook Alert	AutoCall, Offhook Alert	*(continued from previous page)* Both functions apply on a channel-by-channel basis. It would not be appropriate for either of these functions to be applied to a channel that serves in a pool of available channels for general phone traffic. Either function requires an entry in the Outgoing phonebook of the local MultiVOIP and a matched setting in the Inbound Phonebook of the remote voip.
Generate Local Dial Tone	Y/N	*Used for AutoCall only*. If selected, dial tone will be generated locally while the call is being established between gateways. The capability to generate dial tone locally would be particularly useful when there is a lengthy network delay.

Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
AutoCall/Offhook Alert Parameters		
Offhook Alert Timer	0 – 3000 seconds	The length of time that must elapse before the offhook alert is triggered and a call is automatically made to the phone number listed in the **Phone Number** field.
Phone Number	--	Phone number used for Auto Call function or Offhook Alert Timer function. This phone number must correspond to an entry in the Outbound Phonebook of the local MultiVOIP and in the Inbound Phonebook of the remote MultiVOIP (unless a gatekeeper unit is used in the voip system).

Voice/Fax Parameter Definitions (cont'd))		
Field Name	**Values**	**Description**
Dynamic Jitter		
Dynamic Jitter Buffer		**Dynamic Jitter** defines a minimum and a maximum jitter value for voice communications. When receiving voice packets from a remote MultiVOIP, varying delays between packets may occur due to network traffic problems. This is called Jitter. To compensate, the MultiVOIP uses a Dynamic Jitter Buffer. The Jitter Buffer enables the MultiVOIP to wait for delayed voice packets by automatically adjusting the length of the Jitter Buffer between configurable minimum and maximum values. An Optimization Factor adjustment controls how quickly the length of the Jitter Buffer is increased when jitter increases on the network. The length of the jitter buffer directly effects the voice delay between MultiVOIP gateways.
Minimum Jitter Value	60 to 400 ms	The minimum dynamic jitter buffer of **60** milliseconds is the minimum delay that would be acceptable over a low jitter network. Default = 150 msec

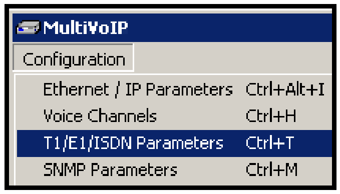
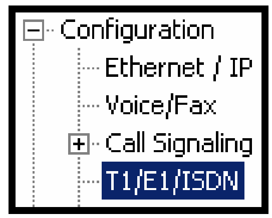
Voice/Fax Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Dynamic Jitter		
Maximum Jitter Value	60 to 400 ms	The maximum dynamic jitter buffer of **400** milliseconds is the maximum delay tolerable over a high jitter network. Default = 300 msec
Optimizat-ion Factor	0 to 12	The **Optimization Factor** determines how quickly the length of the Dynamic Jitter Buffer is changed based on actual jitter encountered on the network. Selecting the minimum value of 0 means low voice delay is desired, but increases the possibility of jitter-induced voice quality problems. Selecting the maximum value of 12 means highest voice quality under jitter conditions is desired at the cost of increased voice delay. Default = **7**.

Modem Relay

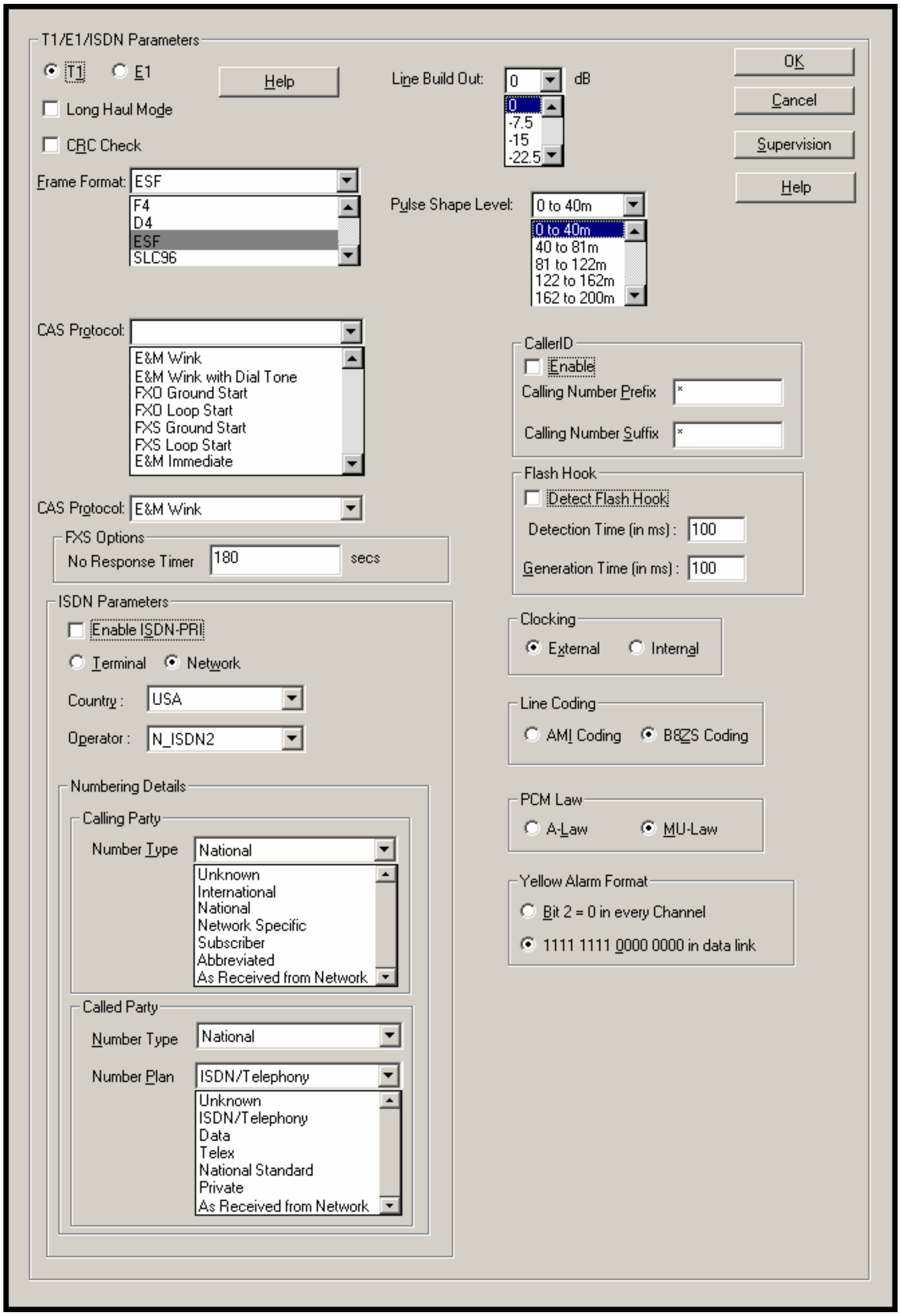
To place modem traffic onto the voip network (an application called "modem relay"), use Coder G.711 mu-law at 64kbps.

Voice/Fax Parameter Definitions (cont'd))		
Field Name	**Values**	**Description**
Auto Disconnect		
Automatic Disconnect-ion	--	The **Automatic Disconnection** group provides four options which can be used singly or in any combination.
Jitter Value	1-65535 milli-seconds	The **Jitter Value** defines the average inter-arrival packet deviation (in milliseconds) before the call is automatically disconnected. The default is **300** milliseconds. A higher value means voice transmission will be more accepting of jitter. A lower value is less tolerant of jitter. Inactive by default. When active, default = 300 ms. However, value must equal or exceed Dynamic Minimum Jitter Value.
Call Duration	1-65535 seconds	**Call Duration** defines the maximum length of time (in seconds) that a call remains connected before the call is automatically disconnected. Inactive by default. When active, default = **180** sec. This may be too short for most configurations, requiring upward adjustment.
Consecutive Packets Lost	1-65535	**Consecutive Packets Lost** defines the number of consecutive packets that are lost after which the call is automatically disconnected. Inactive by default. When active, default = **30**
Network Discon-nection	1 to 65535 seconds; Default = 30 sec.	Specifies how long to wait before disconnecting the call when IP network connectivity with the remote site has been lost.

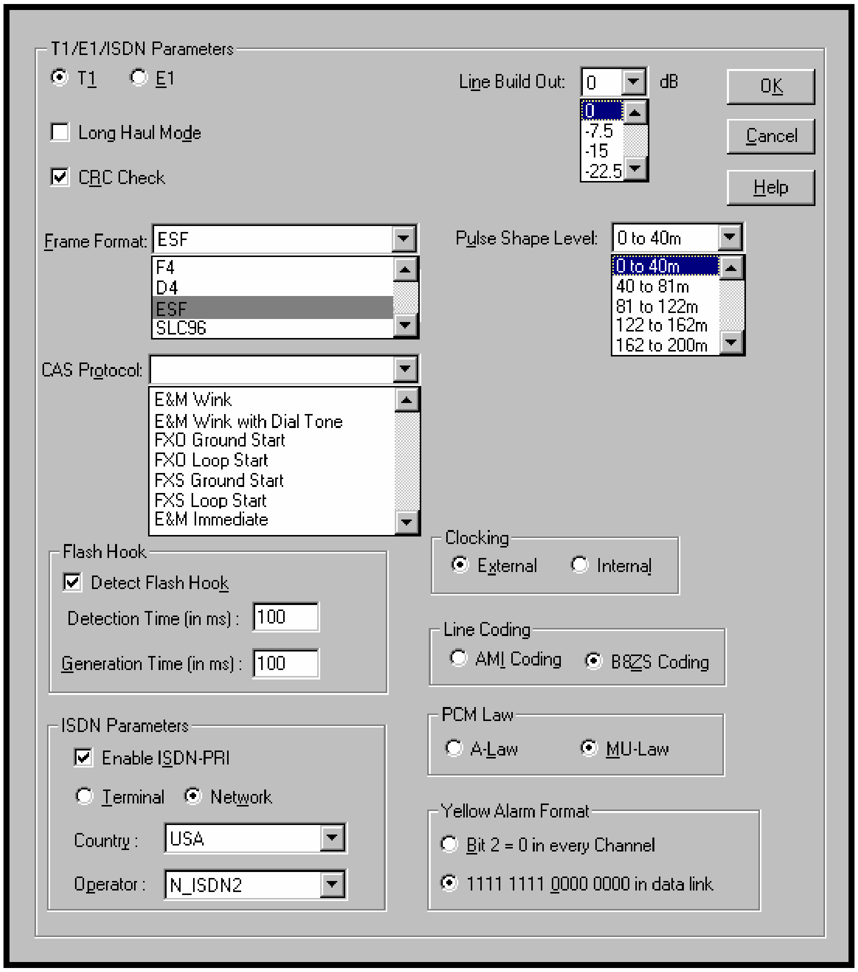
9. **Set T1/E1/ISDN Parameters.** This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

Accessing "T1/E1/ISDN Parameters"	
Pulldown	**Icon**
	--
Shortcut	**Sidebar**
`Ctrl + T`	

In each field, enter the values that fit your particular network.

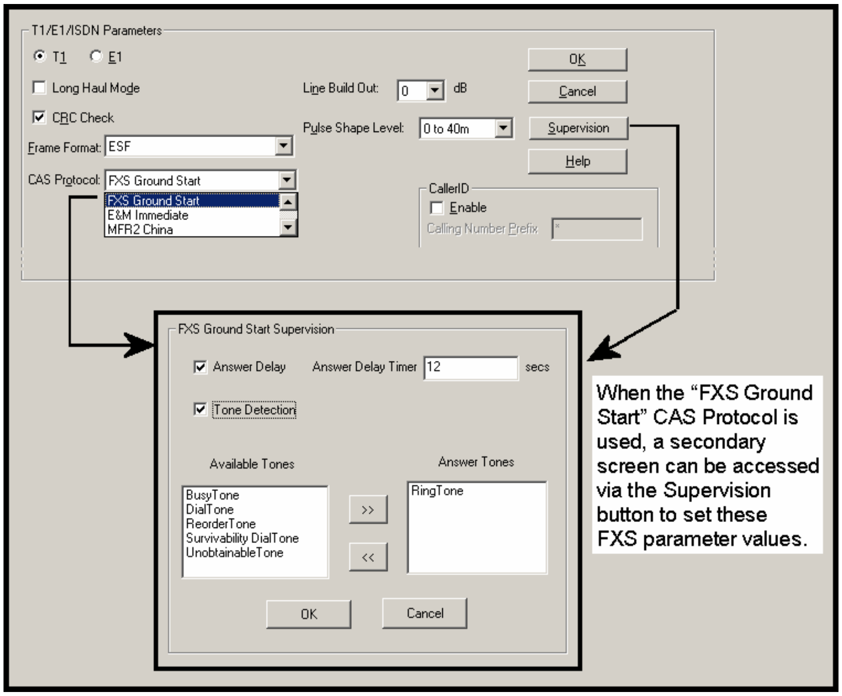


T1 Parameters. The parameters applicable to T1 and their values are shown in the figure below. These **T1 Parameter** fields are described in the tables that follow.



T1 Parameter Definitions		
Field Name	**Values**	**Description**
T1/E1/ISDN	T1	North American digital telephony standard.
Long-Haul Mode	Y/N	In Long-Haul Mode, the MultiVOIP automatically recovers received signals as low as –36 dB. The maximum reachable length with 22 AWG cable is 2000 meters. When Long-Haul Mode is disabled, signals as low as –10 dB can be received. Default: disabled.
CRC Check (Cyclic Redundancy Check)	Y/N	When enabled, allows generation and checking of CRC bits. If not enabled, all check bits in the transmit direction are set. Only applies to ESF frame format. Default: enabled.
Frame Format	F4, D4, ESF, SLC96	Frame Format of MultiVOIP should match that used by PBX or telco. ESF and D4 are commonly used.

T1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
CAS Protocol	E&M Immed Strt E&M Wink Start E&M Wink with dial tone FXO Ground Strt FXO Loop Start FXS Ground Strt FXS Loop Start	Channel Associated Signaling (CAS) is a method of incorporating telephony signaling info into a T1 voice/data stream. In CAS, the signaling bits (the A, B, C, and D bits) are multiplexed into the signal stream of each T1 channel. (By contrast, in Common Channel Signaling (CCS), one channel handles signaling for all other channels.) Each CAS protocol defines the states of the signaling bits during the various stages of a call (IDLE, SEIZED, ANSWER, RING-ON, RING-OFF). The CAS protocol code allows the VOIP to interact properly with the PBX or central-office switch that it serves. If a user has an old MultiVOIP unit (with a firmware version lower than 4.08), and wants to upgrade to 4.08, the latest CAS file (4.08) should also be downloaded into that MultiVOIP unit. The new CAS file ensures proper operation between the MultiVOIP and a PBX. Match this parameter to the setting of PBX or central-office switch.
FXS Options – No Response Timer	1 – 65535 (in seconds)	Length of time before call connection attempt is abandoned. Applicable only when FXS CAS protocol is selected.



T1 Parameter Definitions		
Field Name	**Values**	**Description**
FXS Ground Start Supervision Parameters		
Answer Delay (Enable)	Y/N	When this option is selected, the FXS interface sends the connection notice to the calling party only when the Answer Delay Timer expires. The connection notice is sent regardless of whether or not the called extension has gone offhook.
Answer Delay Timer	numeric (in seconds)	When Answer Delay is enabled, this value determines when the FXS interface sends the connection notice.

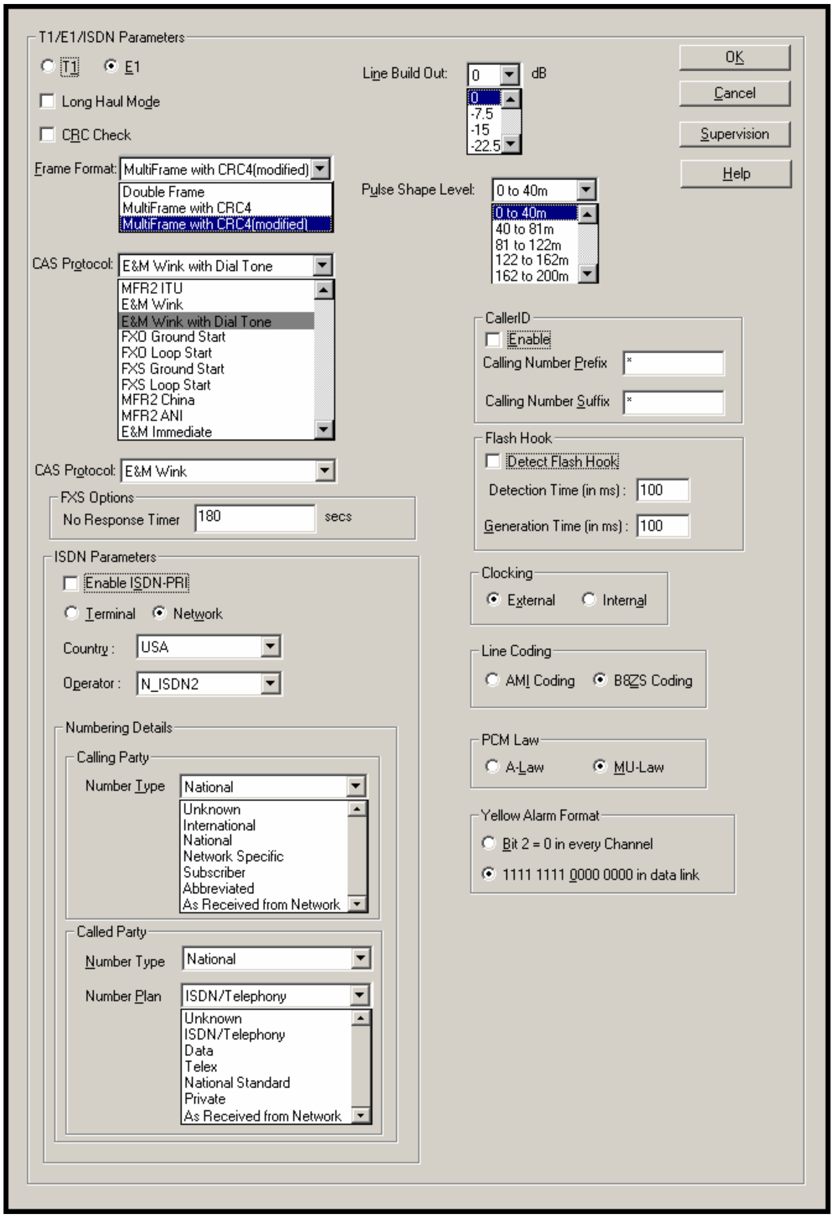
T1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
FXS Ground Start Supervision Parameters		
Tone Detection (Enable)	Y/N	After a specified tone (chosen from the Available Tones list) coming from the PBX is stopped, the FXS interface will send the 'connect' signal to the calling party.
Available Tones (List)	Busy Tone, Dial Tone, Reorder Tone Survivability Dial Tone, Unobtainable Tone	List from which tones can be chosen to signal call answer.
Answer Tones (List)	Busy Tone, Dial Tone, Reorder Tone Survivability Dial Tone, Unobtainable Tone	Currently chosen call-answer supervision tone.
ISDN Parameters		
Field Name	**Values**	**Description**
Enable ISDN-PRI	Y/N	If digital connection is ISDN-PRI type, this box should be checked. When ISDN is enabled, the "CAS Protocols" field is grayed out (ISDN has its own signaling method).
Terminal/ Network	either "Terminal" or "Network"	When "Terminal" is selected, it indicates that the MultiVOIP should emulate the subscriber (terminal) side of the digital connection. When "Network" is selected, it indicates that the MultiVOIP should emulate the central office (network) side of the digital connection. Setting used for MultiVOIP must be opposite to the setting used in the PBX. For example, if the PBX is set to "Terminal," then the MultiVOIP must be set to "Network."

T1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
ISDN Parameters		
Country	see table, later this chapter	Country in which MultiVOIP is operating with ISDN.
Operator	see table, later this chapter	Indicates phone switch manufacturer/model or refers to telco so as to specify the switching system in question. ISDN is implemented somewhat differently in different switches.
Note on Country & Operator options.	—	[ISDN implementation options are shown, arranged by country, in a table below – soon after E1 Parameter Definitions.]
Numbering Details Parameters		
Calling Party Number Type	unknown, national, international, network specific, subscriber, abbreviated, as received from network	Calling party type is part of calling party Number Information element that is sent on ISDN line. The Calling party number information element identifies the origin of a call.
Called Party Number Type	unknown, national, international, network specific, subscriber, abbreviated, as received from network	Called Party Number Type and Called Party Number Plan are part of Calling Party Number Information element that is sent on ISDN line. The Called party number information element identifies destination of a call.
Called Party Number Plan	unknown, ISDN telephony, data, telex, national standard, private, as received from network	The call dialing plan under which the called party operates.

T1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
General T1/E1/ISDN Parameters		
Line Build Out	0 dB, -7.5 dB, -15 dB, -22.5 dB	To reduce the crosstalk on received signals, a transmit attenuator can be placed in the data path. Transmit attenuation is selectable. **Default: O dB**
Pulse Shape Level	0 to 40 Meters 40 to 81 m 81 to 122 m 122 to 162 m 162 to 200 m	Refers to length of cable between MultiVOIP and PBX/telco in meters. Most common will be 0 to 40m.
Caller ID Parameters		
Caller ID Enable	Y/N	Turns Caller ID feature on (if checked) and off (if unchecked).
Calling Number Prefix (Caller ID)	0-9, *, #	A DTMF symbol used to mark the beginning of the calling party number for use with Caller ID. Maximum length: 4 characters.
Calling Number Suffix (Caller ID)	0-9, *, #	A DTMF symbol used to mark the end of the calling party number for use with Caller ID. Maximum length: 4 characters.
Detect Flash Hook	Y/N	This setting determines whether or not the MultiVOIP responds to hook-flash signals.
Detection Time	100 – 1500 milliseconds	Minimum hook-flash time that will be interpreted as a valid flash by the MultiVOIP.
Generation Time	100 – 1500 milliseconds	In some systems, a MultiVOIP might receive a hook-flash signal from an upstream device (a PBX, voip or other device) and must replicate it to a downstream device. This parameter determines the duration of the hook-flash signal that is passed to a downstream device.
Clocking	External/Internal	Set opposite to telco/PBX setting. Example: if telco clocking internal, set VOIP clocking as external.

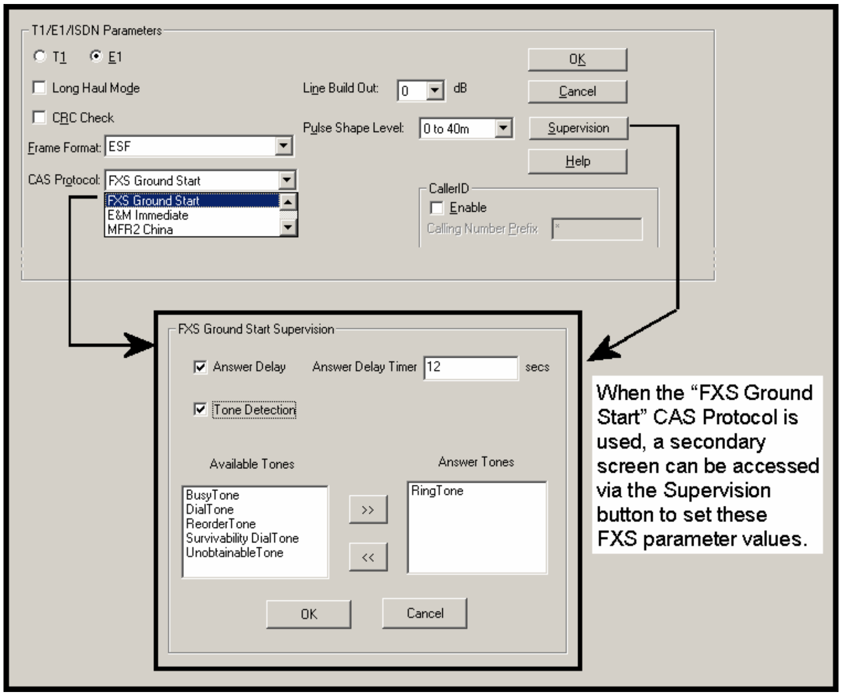
T1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Line Coding	AMI / B8ZS	Match to PBX or telco.
PCM Law	A-Law/Mu-Law	Match to PBX or telco. " Mu-law" is analog-to-digital compression/expansion standard used in North America. "A-law" is European standard.
Yellow Alarm Format	Bit 2 / 1111…	Depending on the Frame Format used, there are choices of Yellow Alarm format, as follows: **D4**: **-Bit2** = 0 in every speech channel **-FS** bit of frame 12 is forced to one. **ESF**: **-Bit2** = 0 in every speech channel **–1111111100000000** pattern in data link channel. Check with your PBX/telco administrator for the correct setting or use the default value (1111 …).

E1 Parameters. The parameters applicable to E1 and their values are shown in the figure below. These **E1 Parameter** fields are described in the tables that follow.



E1 Parameter Definitions		
Field Name	**Values**	**Description**
T1/E1/ISDN	E1	European standard.
Long-Haul Mode	Y/N	In Long-Haul Mode, the MultiVOIP automatically recovers received signals as low as –36 dB. The maximum reachable length with 22 AWG cable is 2000 meters. When Long-Haul Mode is disabled, signals as low as –10 dB can be received. Default: disabled.
CRC Check (Cyclic Redundancy Check)	--	Not applicable to E1.
Frame Format	Double Frame; MultiFrame (with CRC4); MultiFrame (w/CRC4, modified)	Frame Format of MultiVOIP should match that used by PBX or telco.

E1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
CAS Protocol	E&M Immed Strt E&M Wink Start E&M Wink with dial tone FXO Ground Strt FXO Loop Start FXS Ground Strt FXS Loop Start MFR2ITU MFR2 China MFR2 ANI	Channel Associated Signaling (CAS) is a method of incorporating telephony signaling info into an E1 voice/data stream. In CAS, the signaling bits (the A, B, C, and D bits) are multiplexed into the signal stream of each E1 channel. (By contrast, in Common Channel Signaling (CCS), one channel handles signaling for all other channels.) Each CAS protocol defines the states of the signaling bits during the various stages of a call (IDLE, SEIZED, ANSWER, RING-ON, RING-OFF). The CAS protocol code allows the VOIP to interact properly with the PBX or central-office switch that it serves. The need to download CAS protocols arises for only a small minority of VOIP users, and only when PBX/switch is found to be incompatible with standard protocols. Match this parameter to the setting of PBX or central-office switch.
FXS Options – No Response Timer	1 – 65535 (in seconds)	Length of time before call connection attempt is abandoned. Applicable only when FXS is selected as CAS protocol.



E1 Parameter Definitions		
Field Name	**Values**	**Description**
FXS Ground Start Supervision Parameters		
Answer Delay (Enable)	Y/N	When this option is selected, the FXS interface sends the connection notice to the calling party only when the Answer Delay Timer expires. The connection notice is sent regardless of whether or not the called extension has gone offhook.
Answer Delay Timer	numeric (in seconds)	When Answer Delay is enabled, this value determines when the FXS interface sends the connection notice.

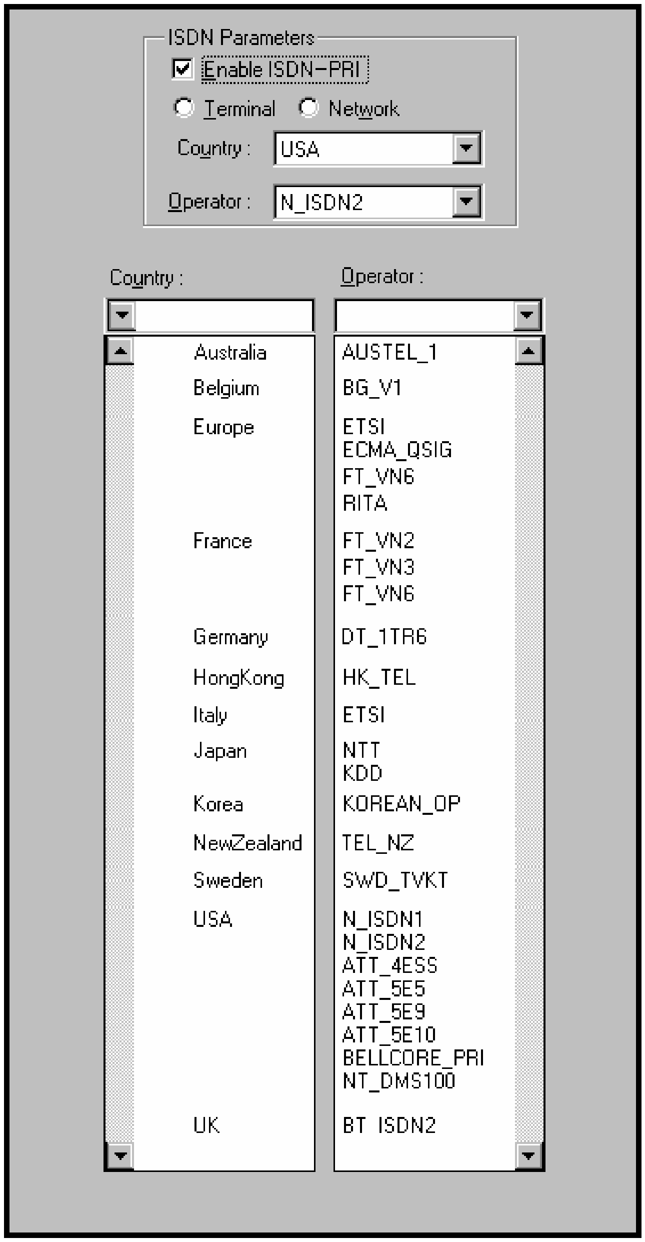
E1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
FXS Ground Start Supervision Parameters		
Tone Detection (Enable)	Y/N	After a specified tone (chosen from the Available Tones list) coming from the PBX is stopped, the FXS interface will send the 'connect' signal to the calling party.
Available Tones (List)	Busy Tone, Dial Tone, Reorder Tone Survivability Dial Tone, Unobtainable Tone	List from which tones can be chosen to signal call answer.
Answer Tones (List)	Busy Tone, Dial Tone, Reorder Tone Survivability Dial Tone, Unobtainable Tone	Currently chosen call-answer supervision tone.
ISDN Parameters		
Field Name	**Values**	**Description**
Enable ISDN-PRI	Y/N	If digital connection is ISDN-PRI type, this box should be checked. When ISDN is enabled, the "CAS Protocols" field is grayed out (ISDN has its own signaling method).
Terminal/ Network	either "Terminal" or "Network"	When "Terminal" is selected, it indicates that the MultiVOIP should emulate the subscriber (terminal) side of the digital connection. When "Network" is selected, it indicates that the MultiVOIP should emulate the central office (network) side of the digital connection. Setting used for MultiVOIP must be opposite to the setting used in the PBX. For example, if the PBX is set to "Terminal," then the MultiVOIP must be set to "Network."

E1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
ISDN Parameters		
Country	see table, later this chapter	Country in which MultiVOIP is operating with ISDN.
Operator	see table, later this chapter	Indicates phone switch manufacturer/model or refers to telco so as to specify the switching system in question. ISDN is implemented somewhat differently in different switches.
Note on Country & Operator options.	—	[ISDN implementation options are shown, arranged by country, in a table below – soon after E1 Parameter Definitions.]
Numbering Details Parameters		
Calling Party Number Type	unknown, national, international, network specific, subscriber, abbreviated, as received from network	Calling party type is part of calling party Number Information element that is sent on ISDN line. The Calling party number information element identifies the origin of a call.
Called Party Number Type	unknown, national, international, network specific, subscriber, abbreviated, as received from network	Called Party Number Type and Called Party Number Plan are part of Calling Party Number Information element that is sent on ISDN line. The Called party number information element identifies destination of a call.
Called Party Number Plan	unknown, ISDN telephony, data, telex, national standard, private, as received from network	The call dialing plan under which the called party operates.

E1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
General E1/E1/ISDN Parameters		
Line Build Out	0 dB, -7.5 dB, -15 dB, -22.5 dB	To reduce the crosstalk on received signals, a transmit attenuator can be placed in the data path. Transmit attenuation is selectable. **Default: O dB**
Pulse Shape Level	0 to 40 Meters 40 to 81 m 81 to 122 m 122 to 162 m 162 to 200 m	Refers to length of cable between MultiVOIP and PBX/telco in meters. Most common will be 0 to 40m.
Caller ID Parameters		
Caller ID Enable	Y/N	Turns Caller ID feature on (if checked) and off (if unchecked).
Calling Number Prefix (Caller ID)	0-9, *, #	A DTMF symbol used to mark the beginning of the calling party number for use with Caller ID. Maximum length: 4 characters.
Calling Number Suffix (Caller ID)	0-9, *, #	A DTMF symbol used to mark the end of the calling party number for use with Caller ID. Maximum length: 4 characters.
Detect Flash Hook	Y/N	This setting determines whether or not the MultiVOIP responds to hook-flash signals.
Detection Time	100 – 1500 milliseconds	Minimum hook-flash time that will be interpreted as a valid flash by the MultiVOIP.
Generation Time	100 – 1500 milliseconds	In some systems, a MultiVOIP might receive a hook-flash signal from an upstream device (a PBX, voip or other device) and must replicate it to a downstream device. This parameter determines the duration of the hook-flash signal that is passed to a downstream device.
Clocking	External/Internal	Set opposite to telco/PBX setting. Example: if telco clocking internal, set VOIP clocking as external.

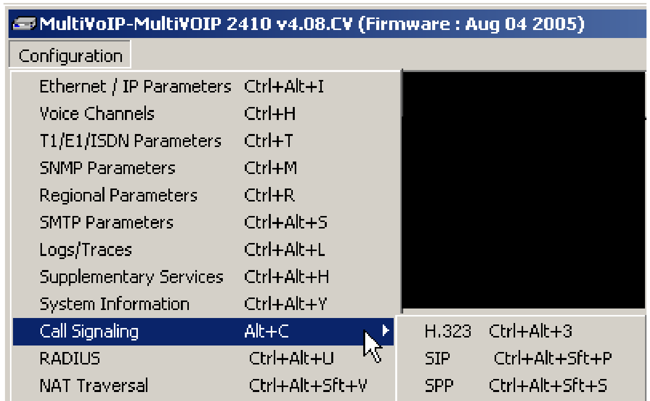
E1 Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Line Coding	AMI / B8ZS	Match to PBX or telco.
PCM Law	A-Law/Mu-Law	Match to PBX or telco. " Mu-law" is analog-to-digital compression/expansion standard used in North America. "A-law" is European standard.
Yellow Alarm Format	Bit 2 / 1111…	Depending on the Frame Format used, there are choices of Yellow Alarm format, as follows: **D4**: **-Bit2** = 0 in every speech channel **-FS** bit of frame 12 is forced to one. **ESF**: **-Bit2** = 0 in every speech channel **–1111111100000000** pattern in data link channel. Check with your PBX/telco administrator for the correct setting or use the default value (1111 …).

10. **Set ISDN Parameters** (if applicable). These parameters are accessible in the **T1/E1/ISDN Parameters** screen. If your T1 or E1 phone line is a Primary Rate Interface ISDN line, enable ISDN-PRI and set it for the particular implementation of ISDN that your telco uses. The ISDN types supported by the digital MultiVOIP units (at press time) are listed below, organized by country.

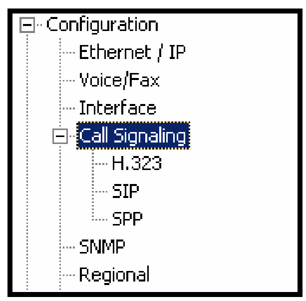


11. **Set Call Signaling Parameters**. This dialog box leads to 3 others, one for each of the call-signaling types supported (H.323, SIP, and SPP). These dialog boxes can be reached by pulldown menu, keyboard shortcut, or a sidebar menu.

<table>
<tr><th colspan="2">Accessing "Call Signaling Parameters"</th></tr>
<tr><th colspan="2">Pulldown</th></tr>
<tr><td colspan="2">

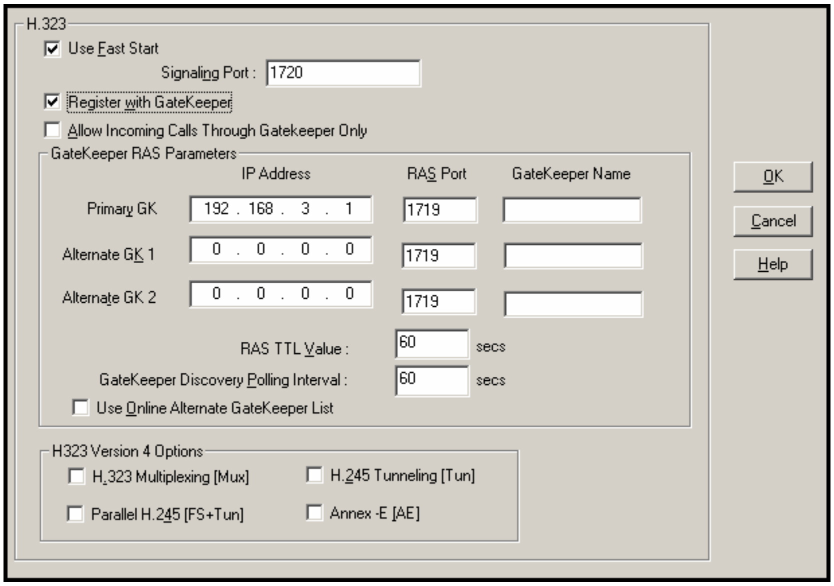


</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Alt + C</td><td>



</td></tr>
</table>

Accessing the Signaling Protocols	
Protocol	
H.323	Ctrl + Alt + 3
SIP	Ctrl + Alt + Shft + P
SPP	Ctrl + Alt + Shft + P



The tables below describes all fields in the general **H.323 Call Signaling** screen.

H.323 Call Signaling Parameter Definitions		
Field Name	**Values**	**Description**
Use Fast Start	Y/N	Enables the H.323 Fast Start procedure. May need to be enabled/disabled for compatibility with third-party VOIP gateways.
Signaling Port	port number	Default: 1720 (H.323)
Register with Gatekeeper	Y/N	Check this field to have traffic on current voip gateway controlled by a gatekeeper.
Allow Incoming Calls Through Gatekeeper Only	Y/N	When selected, incoming calls are accepted only if those calls come through the gatekeeper.

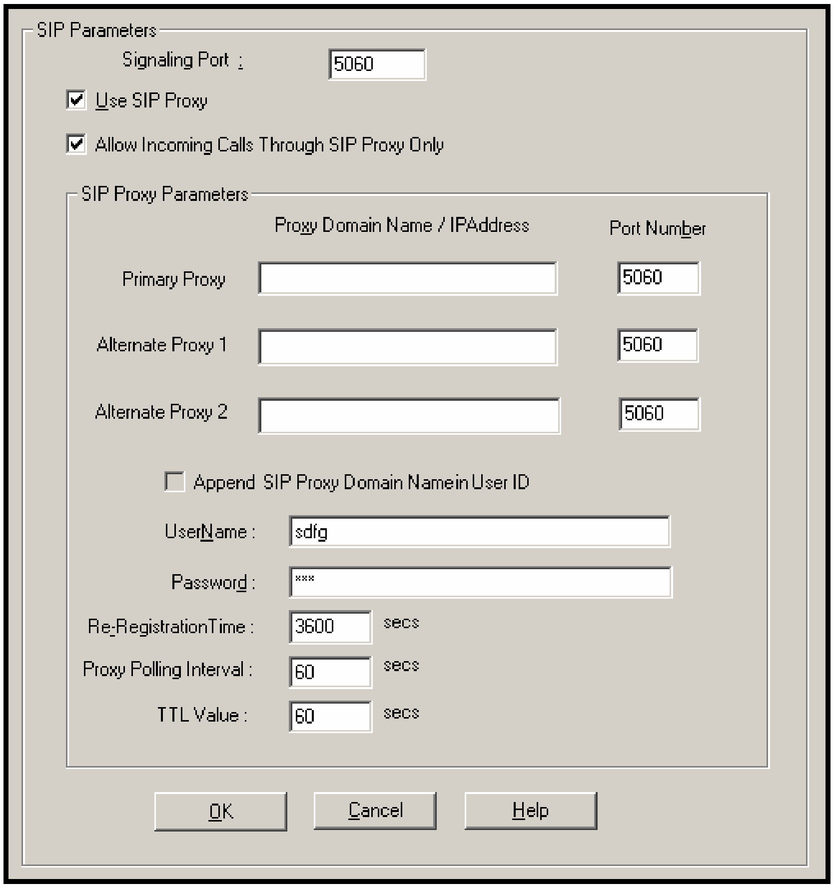
H.323 Call Signaling Parameter Defns (cont'd)		
Field Name	**Values**	**Description**
GateKeeper RAS Parameters		
Primary GK (Gatekeeper)	--	This is the preferred gatekeeper for controlling the traffic of the current voip.
Alternate GK (Gatekeepers) 1 and 2	--	A first and a second alternate gatekeeper can be specified for use by the current voip for situations where the Primary GK is busy or otherwise unavailable.
Gatekeeper / IP Address	n.n.n.n, for n = 0 - 255	IP address of the GateKeeper.
RAS Port	1719	Well-known port number for GateKeepers. Must match port number of GateKeeper, 1719.
Gatekeeper Name	*alpha-numeric string*	Optional. The name of the GateKeeper with which this MultiVOIP is trying to register. A primary gatekeeper and two alternate units are listed.

.

H.323 Call Signaling Parameter Defns (cont'd)		
GateKeeper RAS Parameters		
Field Name	**Values**	**Description**
RAS TTL Value	*in seconds*	The H.323 Gatekeeper "Time to Live" value. As soon as a MultiVOIP gateway registers with a gatekeeper (allowing the gatekeeper to control its call traffic) a countdown timer begins. The RAS TTL Value is the interval of the countdown timer. Before the TTL countdown expires, the MultiVOIP gateway needs to register with the gatekeeper in order to maintain the connection. If the MultiVOIP does not register before the TTL interval expires, the MultiVOIP gateway's registration with the gatekeeper will expire and the gatekeeper will no longer permit call traffic to or from that gateway. Calls in progress will continue to function even if the gateway becomes de-registered.
Gatekeeper Discovery Polling Interval	integer 60 - 300	The interval between the voip gateway's successive attempts to connect to and be governed by a higher level gatekeeper. The Primary GK is the highest level gatekeeper. Alternate GK1 is second; Alternate GK2 is the lowest order gatekeeper.
Use Online Alternate Gatekeeper List (Y/N)	When selected, voip will seek an alternate gatekeeper (when none of the 3 gatekeepers shown on this screen are available) from a list. The list will reside on the Primary gatekeeper or one of the Alternate gatekeepers. The gatekeeper holding the list would download that list onto the voip gateways within the system.	

H.323 Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 Version 4 Parameters		
H.323 Multiplexing (Mux)	Y/N	Signaling for multiple phone calls can be carried on a single port rather than opening a separate signaling port for each call. This conserves bandwidth resources.
H.245 Tunneling (Tun)	Values: Y/N Description: H.245 messages are encapsulated within the Q.931 call-signaling channel. Among other things, the H.245 messages let the two endpoints tell each other what their technical capabilities are and determine who, during the call, will be the client and who the server. Tunneling is the process of transmitting these H.245 messages through the Q.931 channel. The same TCP/IP socket (or logical port) already being used for the Call Signaling Channel is then also used by the H.245 Control Channel. This encapsulation reduces the number of logical ports (sockets) needed and reduces call setup time.	

H.323 Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 Version 4 Parameters		
Parallel H.245 (FS + Tun)	Values: Y/N Description: FS (Fast Start or Fast Connect) is a Q.931 feature of H.323v2 to hasten call setup as well as 'pre-opening' the media channel before the CONNECT message is sent. This pre-opening is a requirement for certain billing activities. Under Parallel H.245 FS + Tun, this Fast Connect feature can operate simultaneously with H.245 Tunneling (see description above).	
Annex –E (AE)	Values: Y/N Description: Multiplexed UDP call signaling transport. Annex E is helpful for high-volume voip system endpoints. Gateways with lesser volume can afford to use TCP to establish calls. However, for larger volume endpoints, the call setup times and system resource usage under TCP can become problematic. Annex E allows endpoints to perform call-signaling functions under the UDP protocol, which involves substantially streamlined overhead. (This feature should not be used on the public Internet because of potential problems with security and bandwidth usage.)	

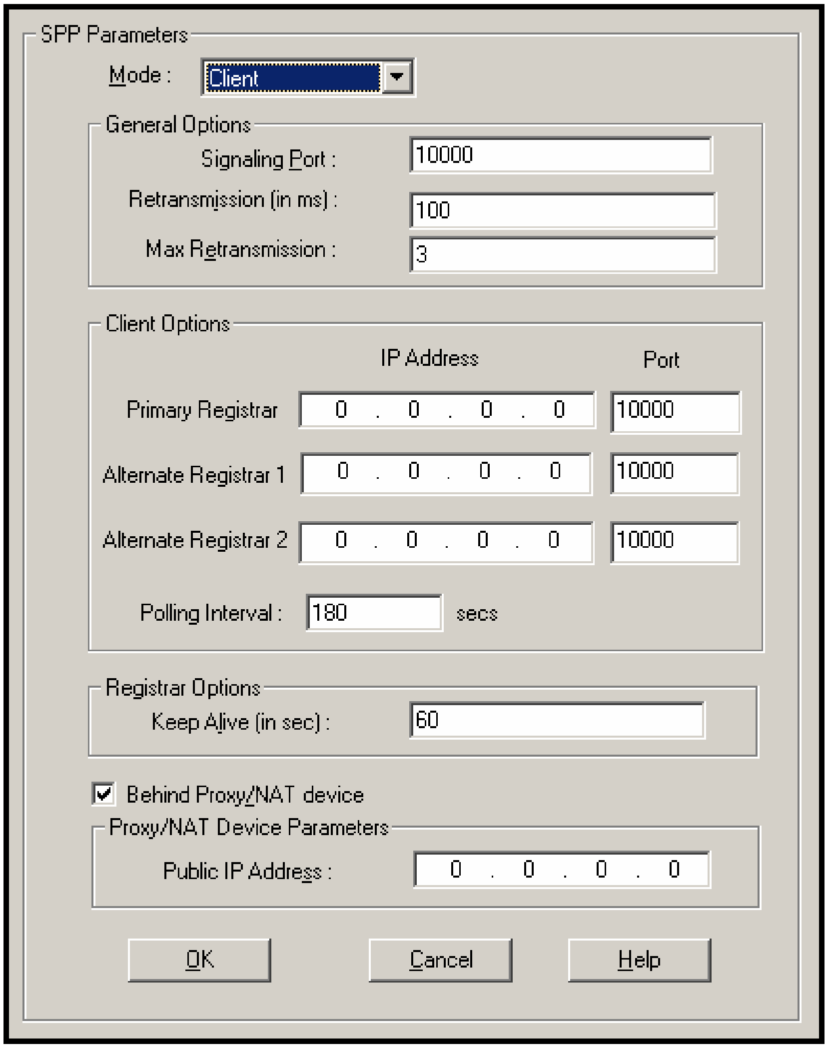


The tables below describes all fields in the general **SIP Call Signaling** screen.

SIP Call Signaling Parameter Definitions		
Field Name	**Values**	**Description**
SIP Proxy Parameters		
Signaling Port		Port number on which the MultiVOIP UserAgent software module will be waiting for any incoming SIP requests.
Use SIP Proxy	Y/N	Allows the MultiVOIP to work in conjunction with a proxy server.

SIP Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
SIP Proxy Parameters		
Allow Incoming Calls Through SIP Proxy Only	Y/N	When selected, incoming calls are accepted only if those calls come through the gatekeeper.
Primary Proxy	--	This is the preferred SIP proxy server for controlling the traffic of the current voip.
Alternate Proxy 1 and 2	--	A first and a second alternate SIP proxy server can be specified for use by the current voip for situations where the Primary proxy server is busy or otherwise unavailable.
Proxy Domain Name / IP Address	n.n.n.n where n=0-255	Network address of the proxy server that the voip is using.
Append SIP Proxy Domain Name in User ID	Y/N	When checked, the domain name of the SIP Proxy serving the MultiVOIP gateway will be included as part of the User ID for that gateway. If unchecked, the SIP Proxy's IP address will be included as part of the User ID instead of the SIP Proxy's domain name.
Port Number		Logical port number for proxy communications.
User Name	Values: alphanumeric Description: Identifier used when proxy server is used in network. If a proxy server is used in a SIP voip network, all clients must enter both a User Name and a Password before being allowed to make a call.	

SIP Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values & Description**	
SIP Proxy Parameters		
Password	Values: alphanumeric Description: Password for proxy server function. See "User Name" description above.	
Re-Registration Time	Values: numeric (in seconds) Description: This is the timeout interval for registration of the MultiVOIP with a SIP proxy server. The time interval begins the moment the MultiVOIP gateway registers with the SIP proxy server and ends at the time specified by the user in the Re-Registration Time field (this field). When/if registration lapses, call traffic routed to/from the MultiVOIP through the SIP proxy server will cease. However, calls in progress will continue to function until they end.	
Proxy Polling Interval	integer 60 - 300	The interval between the voip gateway's successive attempts to connect to and be governed by a higher level SIP proxy server. The Primary Proxy is the highest level gatekeeper. Alternate Proxy 1 is second; Alternate Proxy 2 is the lowest order SIP proxy server.
TTL Value *in seconds*	The SIP proxy "Time to Live" value. As soon as a MultiVOIP gateway registers with a SIP proxy server (allowing the proxy server to control its call traffic) a countdown timer begins. The TTL Value is the interval of the countdown timer. Before the TTL countdown expires, the MultiVOIP gateway needs to register with the gatekeeper in order to maintain the connection. If the MultiVOIP does not register before the TTL interval expires, the MultiVOIP gateway's registration with the proxy server will expire and the proxy server will no longer permit call traffic to or from that gateway. Calls in progress will continue to function even if the gateway becomes de-registered.	


SPP Parameters
Mode :
Client
General Options
Signaling Port :
10000
Retransmission (in ms) :
100
Max Retransmission :
3
Client Options
IP Address
Port
Primary Registrar
0 . 0 . 0 . 0
10000
Alternate Registrar 1
0 . 0 . 0 . 0
10000
Alternate Registrar 2
0 . 0 . 0 . 0
10000
Polling Interval :
180
secs
Registrar Options
Keep Alive (in sec) :
60
Behind Proxy/NAT device
Proxy/NAT Device Parameters
Public IP Address :
0 . 0 . 0 . 0
OK
Cancel
Help

The tables below describes all fields in the general **SPP Call Signaling** screen.

SPP Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Single Port Protocol (SPP)		
Mode	Direct, Client, or Registrar	SPP voip systems can operate in two modes: in the **direct mode**, where all voip gateways have static IP addresses assigned to them; or in the **registrar/client mode**, where one voip gateway serves as registrar and all other gateways, being its clients, point to that registrar. The registrar assigns IP addresses dynamically.
General Options		
Port		The UDP port on which data transmission will occur. Each client voip has its own port. If two client voips are both behind the same firewall, then they must have different ports assigned to them. If there are two clients and each is behind a different firewall, then the clients could have different port numbers or the same port number. (Default port number = 10000.)
Re-trans-mission (in ms)		If packets are lost (as indicated by absence of an acknowledgment) then the endpoint will retransmit the lost packets after this designated time duration has elapsed. (Default value = 2000 milliseconds.)
Max Re-trans-mission		Number of times the voip will re-transmit a lost packet (if no acknowledgment has been received). (Default value = 3)

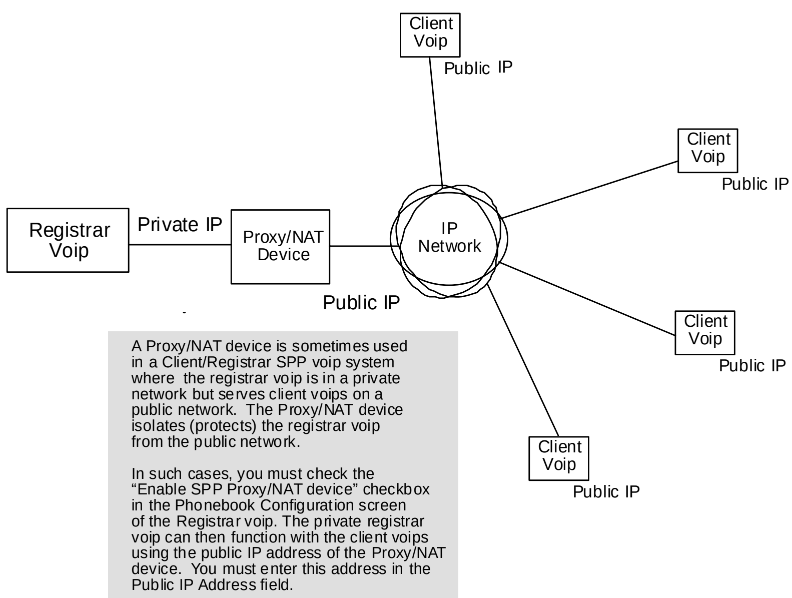
SPP Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Single Port Protocol (SPP) [continued]		
Client Options		Client Option fields are active only in registrar/client mode and only for client voip units.
Primary Registrar	--	This is the preferred SPP registrar gateway for controlling the traffic of the current voip.
Alternate Registrar 1 and 2	--	A first and a second alternate SPP Registrar gateway can be specified for use by the current voip for situations where the Primary Registrar gateway is busy or otherwise unavailable.
Registrar IP Address	n.n.n.n	This is the IP address of the registrar voip to which this client is assigned. (Default value = 0.0.0.0; effectively, there is no useful default value.)
Registrar Port	10000 or other	This is the port number of the registrar voip to which this client is assigned. (Default port number = 10000.)
Polling Interval	integer 60 - 300	The interval between the voip gateway's successive attempts to connect to and be governed by a higher level SPP registrar gateway. The Primary Registrar is the highest level registrar gateway. Alternate Registrar 1 is second; Alternate Registrar 2 is the lowest order SPP registrar gateway.
Registrar Options		Registrar Option fields are active only in registrar/client mode and only for registrar voip units.
Keep Alive (in sec.)	30 – 300 (seconds)	Time-out duration before a registrar will unregister a client that does not send its "I'm here" signal. Client normally sends its "I'm here" signal every 20 seconds. Timeout default = 60 seconds.

SPP Call Signaling Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Proxy/NAT Device Parameters		
Behind Proxy/NAT device	Y/N	Enables MultiVOIP (running in SPP Registrar mode) to operate 'behind' a proxy/NAT device (NAT = Network Address Translation).
Proxy/NAT Device Parameters – Public IP Address	n.n.n.n where n=0-255	The public IP address of the proxy/NAT device which the MultiVOIP is behind.

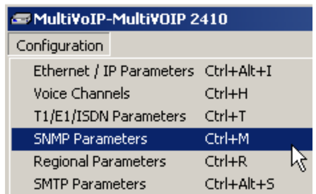
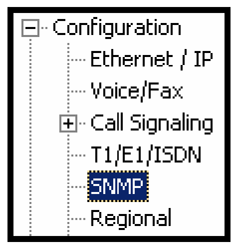
An example of a NAT-equipped SPP network is shown below.

About SPP Proxy/NAT Device Parameters

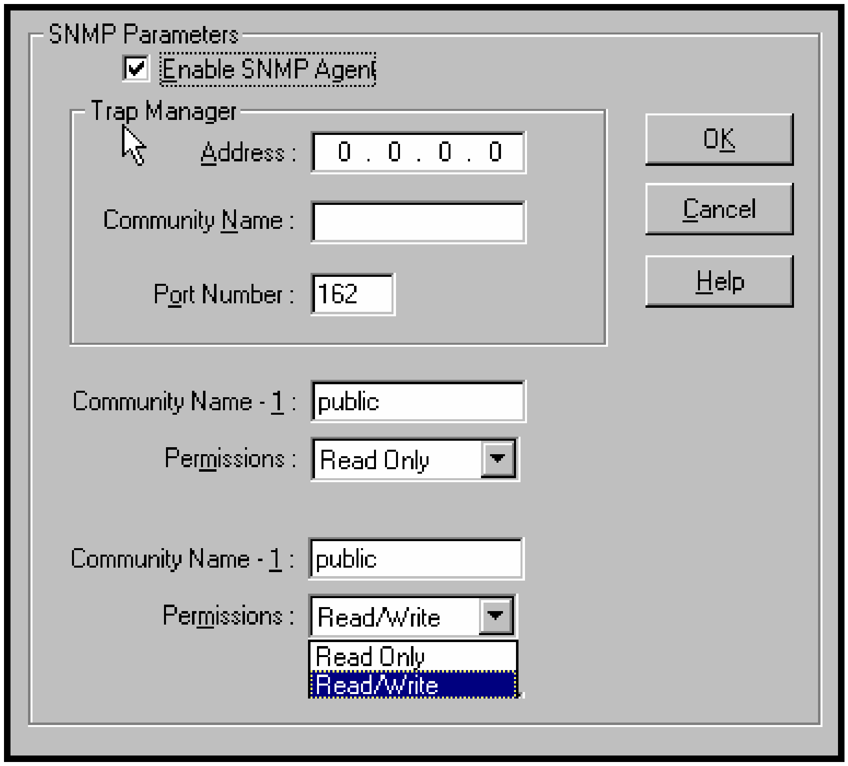
SPP Client/Registrar System



11. **Set SNMP Parameters** (Remote Voip Management). This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar. To make the MultiVOIP controllable by a remote PC running the MultiVoipManager software, check the "Enable SNMP Agent" box on the **SNMP Parameters** screen.

<table>
<tr><th colspan="2">Accessing "SNMP Parameters"</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td></td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td><code>Ctrl + M</code></td><td>
</td></tr>
</table>

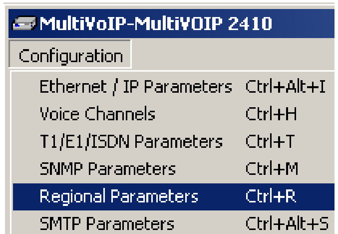
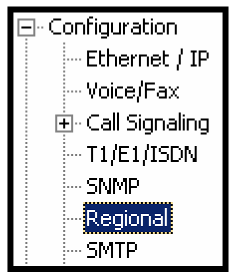
In each field, enter the values that fit your particular system.



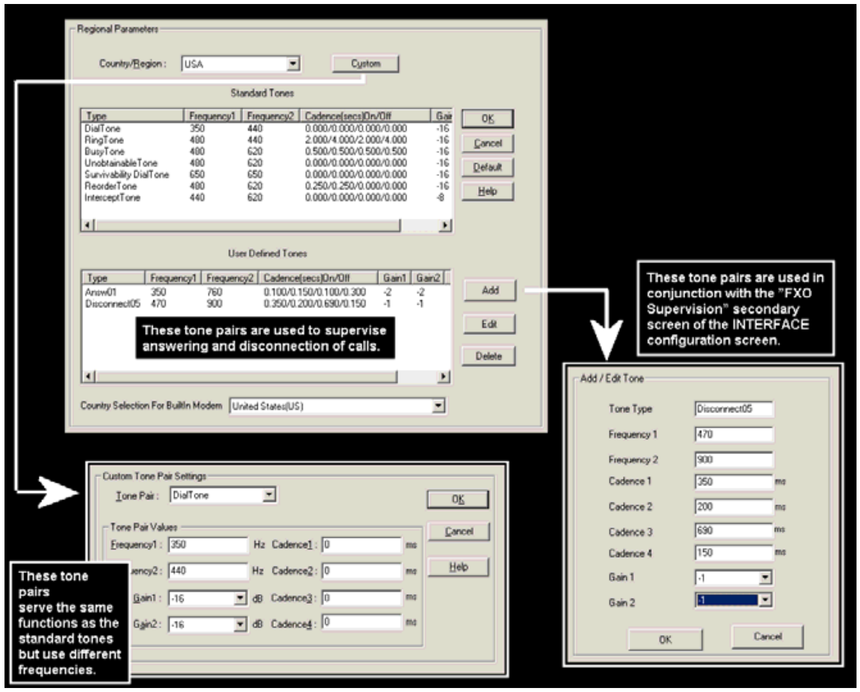
The SNMP Parameter fields are described in the table below.

SNMP Parameter Definitions		
Field Name	**Values**	**Description**
Enable SNMP Agent	Y/N	Enables the SNMP code in the firmware of the MultiVOIP. This must be enabled for the MultiVOIP to communicate with and be controllable by the MultiVoipManager software. Default: disabled
Trap Manager Parameters		
Address	4 places; n.n.n.n n = 0-255	IP address of MultiVoipManager PC.
Community Name	--	A "community" is a group of VOIP endpoints that can communicate with each other. Often "public" is used to designate a grouping where all end users have access to entire VOIP network. However, calling permissions can be configured to restrict access as needed.
Port Number	162	The default port number of the SNMP manager receiving the traps is the standard port 162.
Community Name 1	Length = 19 characters (max.) Case sensitive.	First community grouping.
Permissions	Read-Only, Read/Write	If this community needs to change MultiVOIP settings, select Read/Write. Otherwise, select Read-Only to view settings.
Community Name 2	Length = 19 characters (max.) Case sensitive.	Second community grouping
Permissions	Read-Only, Read/Write	If this community needs to change MultiVOIP settings, select Read/Write. Otherwise, select Read-Only to view settings.

12. **Set Regional Parameters** (Phone Signaling Tones & Cadences). This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing “Regional Parameters”</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td></td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + R</td><td>
</td></tr>
</table>

The **Regional Parameters** screen will appear. For the country selected, the standard set of frequency pairs will be listed for dial tone, busy tone, 'unobtainable' tone (fast busy or trunk busy), ring tone, and other, more specialized tones.

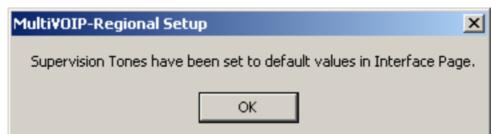


Remote Configuration/Command Modem. Each MVP2410 and MVP3010 MultiVOIP unit contains a built-in modem. This modem allows the MultiVOIP to be configured remotely when a standard POTS line is connected to the "Command Modem" connector on the back panel of the MultiVOIP. In the **Country Selection for Built-In Modem** field (drop-down list), select the country that best fits your situation. This may not be the same as your selection for the **Country/Region** field. The selections in the **Country Selection for Built-In Modem** field entail more detailed groupings of telephony parameters than do the **Country/Region** values.

In each field, enter the values that fit your particular system.

The **Regional Parameters** fields are described in the table below.

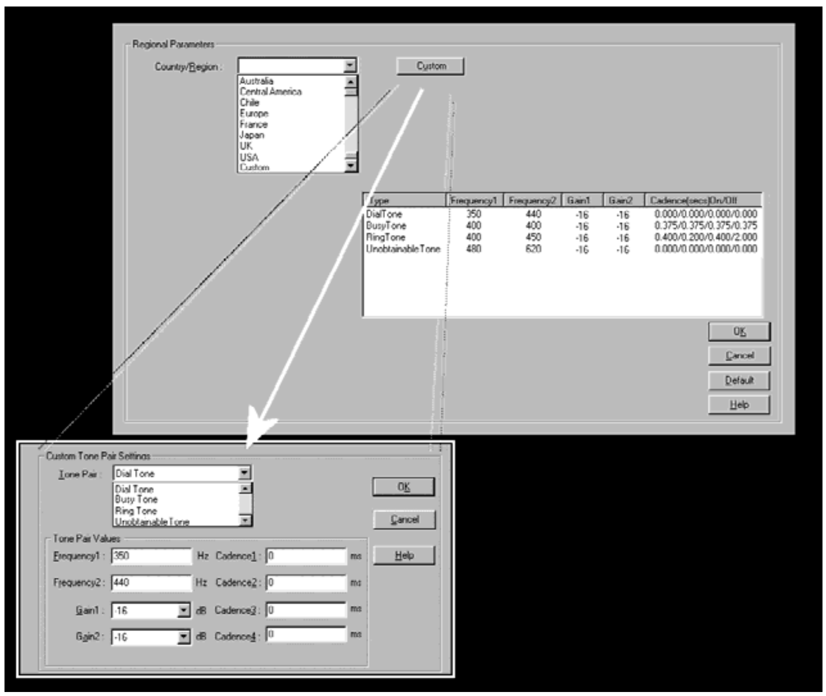
“Regional Parameter” Definitions		
Field Name	**Values**	**Description**
Country/ Region	USA, Japan, UK, Custom	Name of a country or region that uses a certain set of tone pairs for **dial tone, ring tone, busy tone**, **unobtainable tone** (fast busy tone), **survivability tone** (tone heard briefly, 2 seconds, after going offhook denoting survivable mode of VOIP unit), **re-order tone** (a tone pattern indicating the need for the user to hang up the phone), and **intercept tone** (a tone that warns an a party that has gone off hook but has not begun dialing, within a prescribed time, that an automatic emergency or attendant number will be called; the automatic call can be used to direct an attendant’s attention to a disabled or distressed caller, allowing an appropriate response to be made). In some cases, the tone-pair scheme denoted by a country name may also be used outside of that country. The “Custom” option (button) assures that any tone-pairing scheme worldwide can be accommodated. **Note**: **Intercept tone** is applicable only when the FXS telephony interface has been chosen in the **Interface** screen and when the AutoCall / OffHook Alert field is set to OffHook Alert in the **Voice/Fax Parameters** screen. The time allowed for dialing before the automatic calling process begins is set in the Offhook Alert Timer field of the **Voice/Fax Parameters** screen.

“Regional Parameter” Definitions		
Field Name	**Values**	**Description**
Country/ Region	USA, Japan, UK, Custom **Note**: “Survivability” tone indicates a special type of call-routing redundancy & applies to MultiVantage voip units only.	Name of a country or region that uses a certain set of tone pairs for dial tone, ring tone, busy tone, and ‘unobtainable’ tone (fast busy tone), survivability tone (tone heard briefly, 2 seconds, after going offhook denoting survivable mode of voip unit) and re-order tone (a tone pattern indicating the need for the user to hang up the phone). In some cases, the tone-pair scheme denoted by a country name may also be used outside of that country. The “Custom” option (button) assures that any tone-pairing scheme worldwide can be accommodated.
Advisory screen	 This message screen appears whenever the Country field is changed. It informs the operator that, upon change of the Country field value, all User Defined Tones will be deleted.	
Standard Tones fields		
Type column	dial tone, ring tone, busy tone, unobtainable tone (fast busy), survivability tone, re-order tone	Type of telephony tone-pair for which frequency, gain, and cadence are being presented.
Frequency 1	freq. in Hertz	Lower frequency of pair.
Frequency 2	freq. in Hertz	Higher frequency of pair.

"Regional Parameter" Definitions (cont'd)		
Field Name	**Values**	**Description**
Standard Tones fields (cont'd)		
Gain 1	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of lower frequency of pair. This applies to the dial, ring, busy and 'unobtainable' tones that the MultiVOIP outputs as audio to the FXS, FXS, or E&M port. **Default: -16dB**
Gain 2	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of higher frequency of pair. This applies to the dial, ring, busy, and 'unobtainable' (fast busy) tones that the MultiVOIP outputs as audio to the FXS, FXO, or E&M port. **Default: -16dB**
Cadence (msec) On/Off	n/n/n/n four integer time values in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, connection unobtainable (fast busy), dial tone ("0" indicates continuous tone), survivability, and re-order. Default values differ for different countries/regions. Although most cadences have only two parts (an "on" duration and an "off" duration), some telephony cadences have four parts. Most cadences, then, are expressed as two iterations of a two-part sequence. Although this is redundant, it is necessary to allow for expression of 4-part cadences.
Custom (button)	--	Click on the "Custom" button to bring up the **Custom Tone Pair Settings** screen. (The "Custom" button is active only when "Custom" is selected in the **Country/Region** field.) This screen allows the user to specify tone pair attributes that are not found in any of the standard national/regional telephony toning schemes.

"Regional Parameter" Definitions (cont'd)		
Field Name	**Values**	**Description**
Country Selection for Built-In Modem (*not applicable to MVP-130/130FXS MVP210, MVP410ST, or MVP810ST*)	country name	MultiVOIP units operating with the X.06 software release (and above) include a built-in modem. The administrator can dial into this modem to configure the MultiVOIP unit remotely. The country name values in this field set telephony parameters that allow the modem to work in the listed country. This value may be different than the Country/Region value. For example, a user may need to choose "Europe" as the Country/Region value but "Denmark" as the Country-Selection-for-Built-In-Modem value.
User Defined Tones fields		
Type column	alphanumeric name specified by user	Name of supervisory tone pair. Cannot be same as name of any standard tone pair.
Frequency 1	freq. in Hertz	Lower frequency of pair.
Frequency 2	freq. in Hertz	Higher frequency of pair.
Gain 1	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of lower frequency of pair. This applies to any supervisory tones that the MultiVOIP outputs as audio to the FXS, FXS, or E&M port. **Default: -16dB**
Gain 2	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of higher frequency of pair. This applies to any supervisory tones that the MultiVOIP outputs as audio to the FXS, FXO, or E&M port. **Default: -16dB**
Cadence (msec) On/Off	n/n/n/n four integer time values in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote supervisory tones specified by user. Supervisory tones relate to answering and disconnection of calls. Although most cadences have only two parts (an "on" duration and an "off" duration), some telephony cadences have four parts. Most cadences, then, are expressed as two iterations of a two-part sequence. Although this is redundant, it is necessary to allow for expression of 4-part cadences.

13. **Set Custom Tones and Cadences** (optional). The **Regional Parameters** dialog box has a secondary dialog box that allows you to customize DTMF tone pairs to create unique ring-tones, dial-tones, busy-tones or "unobtainable" tones (fast busy signal) or "re-order" tones (telling the user that she must hang up an off-hook phone) or "survivability" tones (an indication of call-routing redundancy) for your system. This screen allows the user to specify tone-pair attributes that are not found in any of the standard national/regional telephony toning schemes. To access this customization feature, click on the **Custom** button on the **Regional Parameters** screen. (The "Custom" button is active only when "Custom" is selected in the **Country/Region** field.)

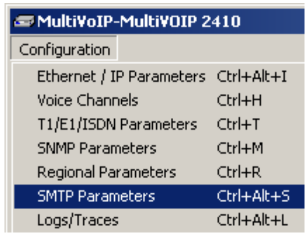
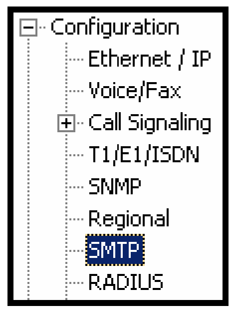


The **Custom Tone-Pair Settings** fields are described in the table below.

Custom Tone-Pair Settings Definitions		
Field Name	**Values**	**Description**
Tone Pair	dial tone, busy tone, ring tone, 'unobtainable' tone, survivability tone, re-order tone	Identifies the type of telephony signaling tone for which frequencies are being specified.
TONE PAIR VALUES		**About Defaults**: US telephony values are used as defaults on this screen. However, since this dialog box is provided to allow custom tone-pair settings, default values are essentially irrelevant.
Frequency 1	frequency in Hertz	Frequency of lower tone of pair. This outbound tone pair enters the MultiVOIP at the input port.
Frequency 2	frequency in Hertz	Frequency of higher tone of pair. This outbound tone pair enters the MultiVOIP at the input port.
Gain 1	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of lower frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the input port. **Default = -16dB**
Gain 2	gain in dB +3dB to –31dB and "mute" setting	Amplification factor of higher frequency of pair. This figure describes amplification that the MultiVOIP applies to outbound tones entering the MultiVOIP at the input port. **Default = -16dB**

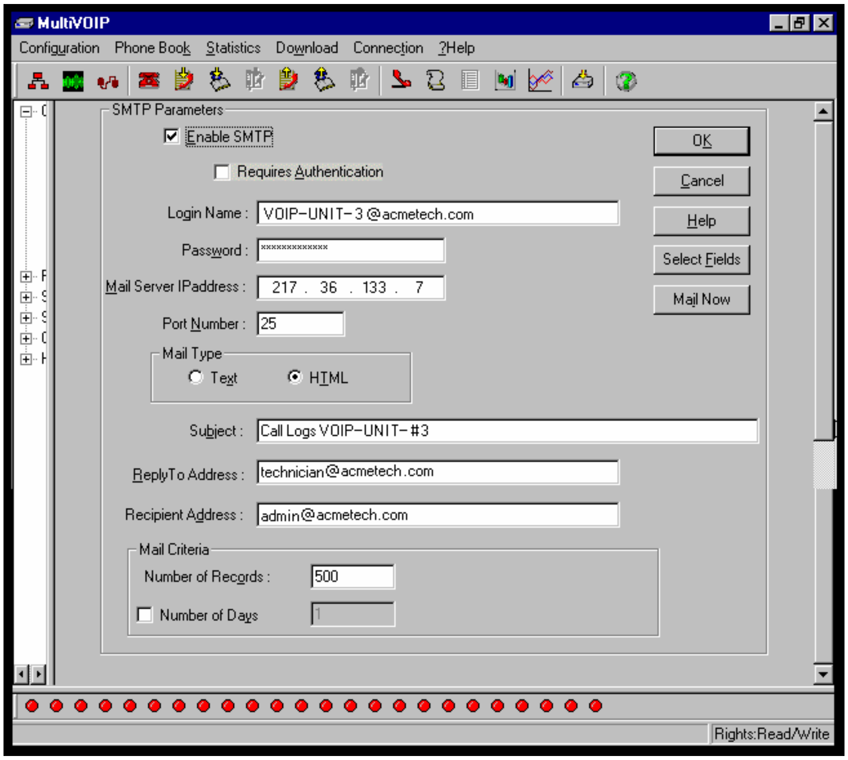
Custom Tone-Pair Settings Definitions		
Field Name	**Values**	**Description**
Cadence 1	integer time value in milli-seconds; zero value for dial-tone indicates continuous tone	On/off pattern of tone durations used to denote phone ringing, phone busy, dial tone ("0" indicates continuous tone) survivability and re-order. Cadence 1 is duration of first period of tone being "on" in the cadence of the telephony signal (which could be ring-tone, busy-tone, unobtainable-tone, or dial tone).
Cadence 2	duration in milliseconds	Cadence 2 is duration of first "off" period in signaling cadence.
Cadence 3	duration in milliseconds	Cadence 3 is duration of second "on" period in signaling cadence.
Cadence 4	duration in milliseconds	Cadence 4 is duration of second "off" period in the signaling cadence, after which the 4-part cadence pattern of the telephony signal repeats.

14. **Set SMTP Parameters** (Log Reports by Email). The **SMTP Parameters** screen is applicable when the VOIP administrator has chosen to receive log reports by email (this is done by selecting the "SMTP" checkbox in the **Others** screen and selecting "Enable SMTP" in the **SMTP Parameters** screen.). The **SMTP Parameters** screen can be reached by pulldown menu, keyboard shortcut, or sidebar.

Accessing "SMTP Parameters"	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + Alt + S`	

MultiVOIP as Email Sender. When SMTP is used, the MultiVOIP will actually be given its own email account (with Login Name and Password) on some mail server connected to the IP network. Using this account, the MultiVOIP will then send out email messages containing log report information. The "Recipient" of the log report email is ordinarily the VoIP administrator. Because the MultiVOIP cannot receive email, a "Reply-To" address must also be set up. Ordinarily, the "Reply-To" address is that of a technician who has access to the mail server or MultiVOIP or both, and the VoIP administrator might also be designated as the "Reply-To" party. The main function of the Reply-To address is to receive error or failure messages regarding the emailed reports.

The **SMTP Parameters** screen is shown below



“SMTP Parameters” Definitions		
Field Name	**Values**	**Description**
Enable SMTP	Y/N	In order to send log reports by email, this box must be checked. However, to enable SMTP functionality, you must also select “SMTP” in the **Logs** screen.
Requires Authentication	Y/N	If this checkbox is checked, the MultiVOIP will send Authentication information to the SMTP server. The authentication information indicates whether or not the email sender has permission to use the SMTP server.
Login Name	alpha-numeric, per email domain	This is the User Name for the MultiVOIP unit’s email account.

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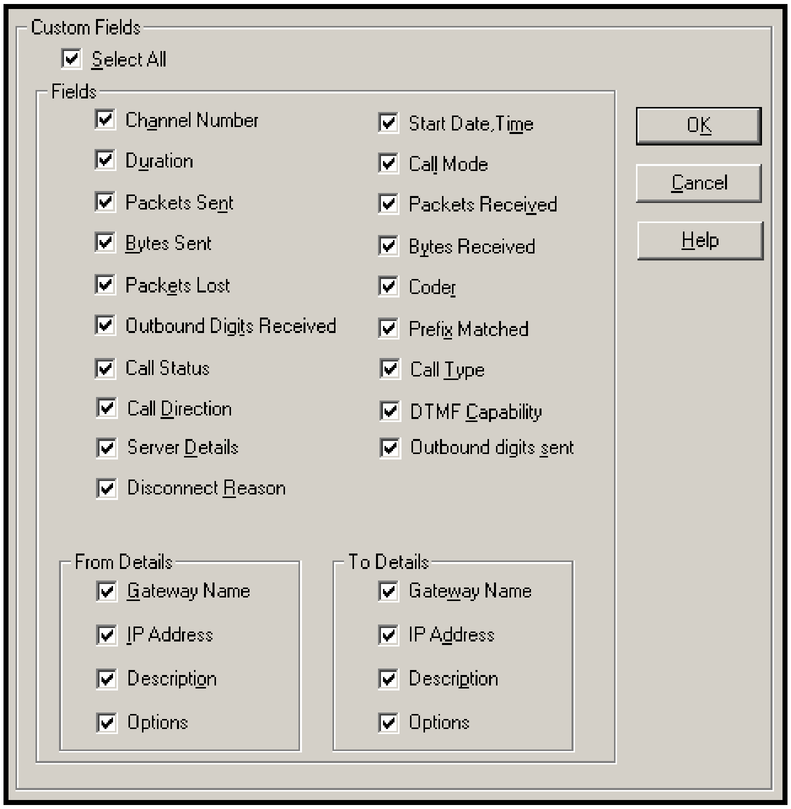
“SMTP Parameters” Definitions (cont’d)		
Field Name	**Values**	**Description**
Password	alpha-numeric	Login password for MultiVOIP unit’s email account.
Mail Server IP Address	n.n.n.n for n= 0 to 255	This is the mail server’s IP address. This mail server must be accessible on the IP network to which the MultiVOIP is connected.
Port Number	25	25 is a standard port number for SMTP.
Mail Type	text or html	Mail type in which log reports will be sent.
Subject	text	User specified. Subject line that will appear for all emailed log reports for this MultiVOIP unit.
Reply-To Address	email address	User specified. This email address functions as a source email identifier for the MultiVOIP, which, of course, cannot usefully receive email messages. The Reply-To address provides a destination for returned messages indicating the status of messages sent by the MultiVOIP (esp. to indicate when log report email was undeliverable or when an error has occurred).
Recipient Address	email address	User specified. Email address at which VOIP administrator will receive log reports.
Mail Criteria		Criteria for sending log summary by email. The log summary email will be sent out either when the user-specified number of log messages has accumulated, or once every day or multiple days, *which ever comes first*.
Number of Records	integer	This is the number of log records that must accumulate to trigger the sending of a log-summary email.
Number of Days	integer	This is the number of days that must pass before triggering the sending of a log-summary email.

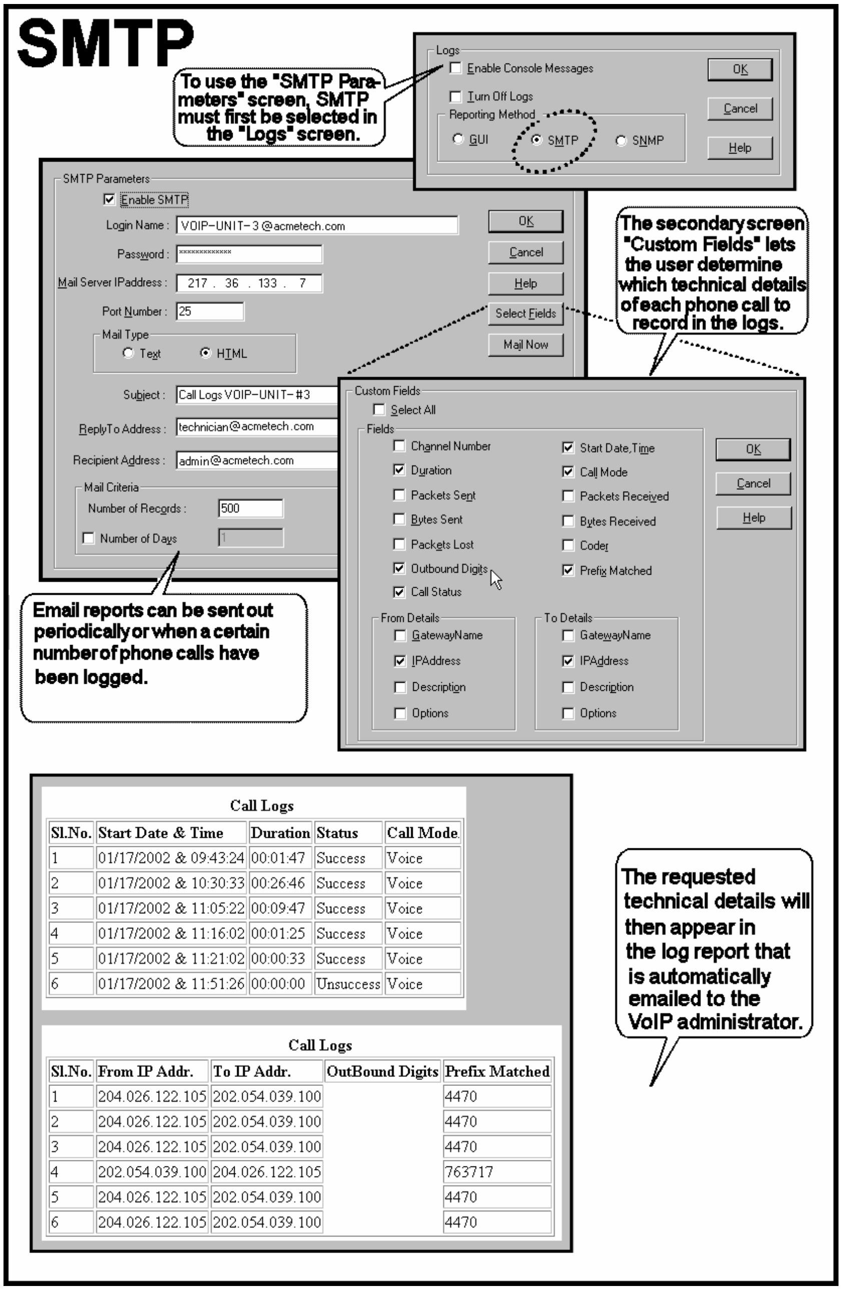
The **SMTP Parameters** dialog box has a secondary dialog box, **Custom Fields**, that allows you to customize email log messages for the MultiVOIP. The MultiVOIP software logs data about many aspects of the call traffic going through the MultiVOIP. The Custom Fields screen lets you pick which aspects will be included in the email log reports.



"Custom Fields" Definitions			
Field	**Description**	**Field**	**Description**
Select All	Log report to include all fields shown.		
Channel Number	Data channel carrying call.	Start Date, Time	Date and time the phone call began.
Duration	Length of call.	Call Mode	Voice or fax.
Packets Sent	Total packets sent in call.	Packets Received	Total packets received in call.

"Custom Fields" Definitions (cont'd)			
Field	**Description**	**Field**	**Description**
Bytes Sent	Total bytes sent in call.	Bytes Received	Total bytes received in call.
Packets Lost	Packets lost in call.	Coder	Voice Coder /Compression Rate used for call will be listed in log.
Outbound Digits Received	The DTMF dialing digits received by this gateway from the remote gateway presuming that DTMF is set to "Out of Band."	Prefix Matched	When selected, the phonebook prefix matched in processing the call will be listed in log.
Call Status	Successful or unsuccessful.	Call Type	Indicates the Call Signaling protocol used for the call (H.323, SIP, or SPP).
Call Direction	Indicates call's originating party.	DTMF Capability	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." The corresponding field values differ for the 3 different voip protocols. For H.323, this field can display "Out of Band" or "Inband". For SIP it can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition. For SPP it can display "Out of Band RFC2833" or "Inband".

"Custom Fields" Definitions (cont'd)			
Field	**Description**	**Field**	**Description**
Server Details	The IP address of the traffic control server (if any) being used (whether an H.323 gatekeeper, a SIP proxy, or an SPP registrar gateway) will be displayed here if the call is handled through that server.	Outbound Digits Sent	The dialing digits sent by this gateway to the remote gateway presuming that DTMF is set to "Out of Band."
Disconnect Reason	Indicates whether the call was disconnected simply because the desired conversation was done or some other irregular cause occasioned disconnection (e.g., a technical error or failure). Values are "Normal" and "Local" disconnection.		
From Details		**To Details**	
Gateway Number	Originating gateway	Gatew N.	Completing or answering gateway
IP Addr	IP address where call originated.	IP Addr	IP address where call was completed or answered.
Descript	Identifier of site where call originated.	Descript	Identifier of site where call was completed or answered.
Options	When selected, log will not Silence Compression and Forward Error Correction by call originator.	Options	When selected, log will not use Silence Compression and Forward Error Correction by party answering call.



Call Logs

Sl.No.	Start Date & Time	Duration	Status	Call Mode
1	01/17/2002 & 09:43:24	00:01:47	Success	Voice
2	01/17/2002 & 10:30:33	00:26:46	Success	Voice
3	01/17/2002 & 11:05:22	00:09:47	Success	Voice
4	01/17/2002 & 11:16:02	00:01:25	Success	Voice
5	01/17/2002 & 11:21:02	00:00:33	Success	Voice
6	01/17/2002 & 11:51:26	00:00:00	Unsuccess	Voice

Call Logs

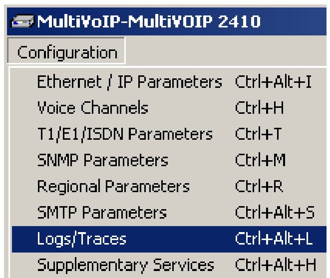
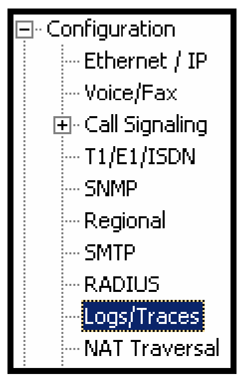
Sl.No.	From IP Addr.	To IP Addr.	OutBound Digits	Prefix Matched
1	204.026.122.105	202.054.039.100		4470
2	204.026.122.105	202.054.039.100		4470
3	204.026.122.105	202.054.039.100		4470
4	202.054.039.100	204.026.122.105		763717
5	204.026.122.105	202.054.039.100		4470
6	204.026.122.105	202.054.039.100		4470

15. **Set Log Reporting Method**. The **Logs** screen lets you choose how the VoIP administrator will receive log reports about the MultiVOIP's performance and the phone call traffic that is passing through it. Log reports can be received in one of three ways:

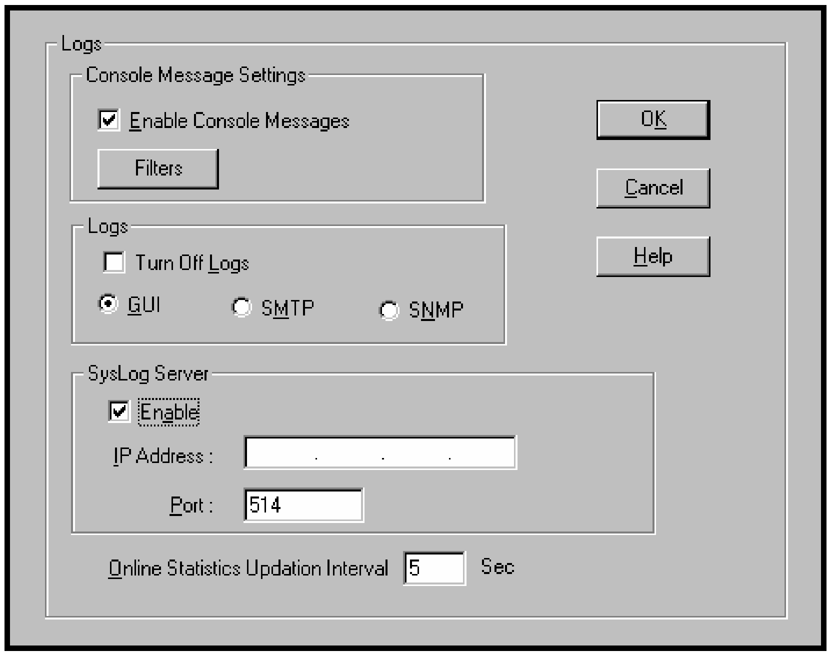
    A. in the MultiVOIP program (GUI),

    B. via email (SMTP), or

    C. at the MultiVoipManager remote voip system management program (SNMP).

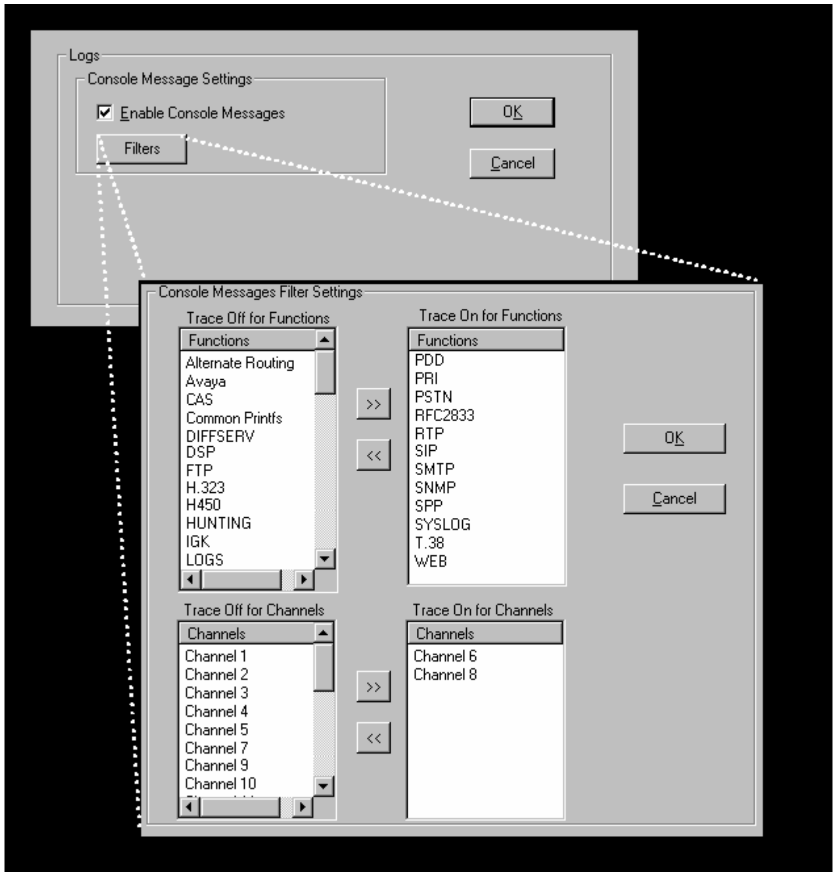
<table>
<tr><th colspan="2">Accessing "Logs/Traces" Screen</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td></td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + Alt + L</td><td>
</td></tr>
</table>

If you enable console messages, you can customize the types of messages to be included/excluded in log reports by clicking on the "Filters" button and using the **Console Messages Filter Settings** screen (see subsequent page). If you use the logging function, select the logging option that applies to your VoIP system design. If you intend to use a SysLog Server program for logging, click in that Enable check box. The common SysLog logical port number is 514. If you intend to use the MultiVOIP web browser GUI for configuration and control of MultiVOIP units, be aware that the web browser GUI does not support logs directly. However, when the web browser GUI is used, log files can still be sent to the voip administrator via email (which requires activating the SMTP logging option in this screen).

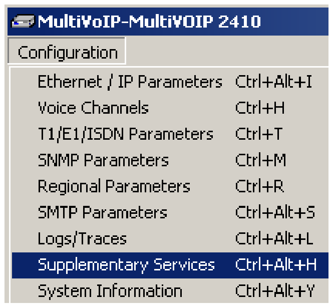
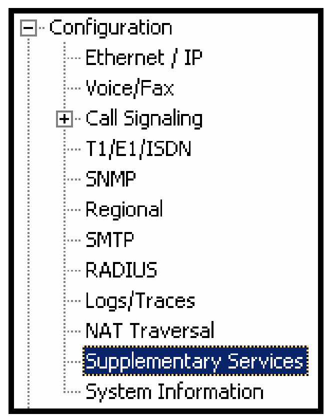


“Logs” Screen Definitions		
Field Name	**Values**	**Description**
Enable Console Messages	Y/N	Allows MultiVOIP debugging messages to be read via a basic terminal program like HyperTerminal ™ or equivalent. Normally, this should be disabled because it uses MultiVOIP processing resources. Console messages are meant for tech support personnel.
Filters (button)		Click to access secondary screen on where console messages can be included/excluded by category and on a per-channel basis. (See the Console Messages Filter Settings screen on subsequent page.)
Turn Off Logs	Y/N	Check to disable log-reporting function.
Logs Buttons		Only one of these three log reporting methods, GUI, SMTP, or SNMP, may be chosen.
GUI	Y/N	User must view logs at the MultiVOIP configuration program.
SNMP	Y/N	Log messages will be delivered to the MultiVoipManager application program.
SMTP	Y/N	Log messages will be sent to user-specified email address.
SysLog Server Enable	Y/N	This box must be checked if logging is to be done in conjunction with a SysLog Server program. For more on SysLog Server, see *Operation & Maintenance* chapter.
IP Address	n.n.n.n for n= 0-255	IP address of computer, connected to voip network, on which SysLog Server program is running.
Port	514	Logical port for SysLog Server. 514 is commonly used.
Online Statistics Updation Interval	integer	Set the interval (in seconds) at which logging information will be updated.

To customize console messages by category and/or by channel, click on "Filters" and use the **Console Messages Filters Settings** screen.

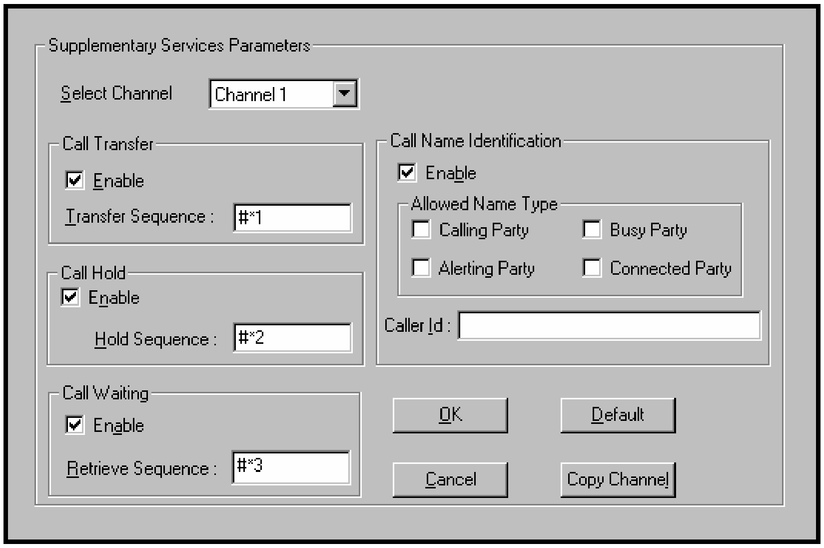


16. **Set Supplementary Services Parameters.** This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing “Supplementary Services” Parameters</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td></td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td><code>Ctrl + Alt +H</code></td><td>
</td></tr>
</table>

Supplementary Services features derive from the H.450 standard, which brings to voip telephony functionality once only available with PSTN or PBX telephony. Supplementary Services features can be used under H.323 only and *not* under SIP. Even though the H.450 standard refers only to H.323, Supplementary Services are still applicable to the SIP and SPP voip protocols.

In each field, enter the values that fit your particular network.



Of the features implemented under Supplementary Services, three are very closely related: Call Transfer, Call Hold, and Call Waiting. Call Name Identification is similar but not identical to the premium PSTN feature commonly known as **Caller ID**.

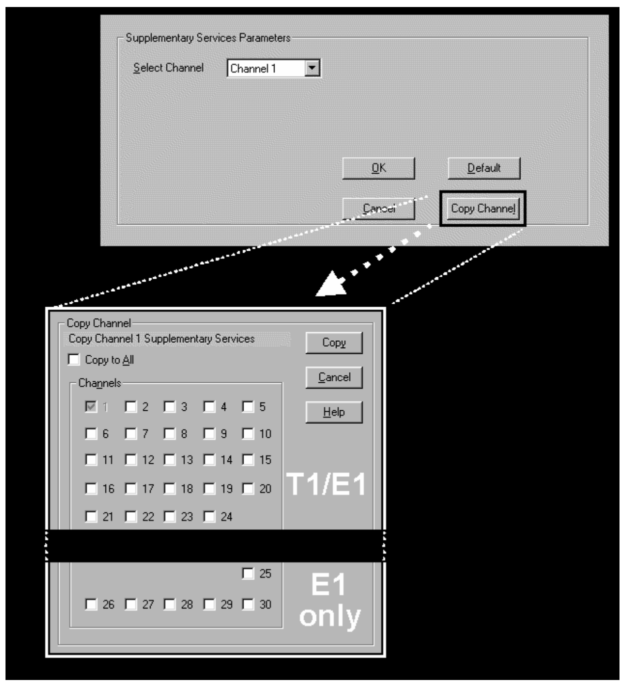
Call Transfer. Call Transfer allows one party to re-connect the party with whom they have been speaking to a third party. The first party is disconnected when the third party becomes connected. Feature is invoked by a programmable phone keypad sequence (for example, #7).

Call Hold. Call Hold allows one party to maintain an idle (non-talking) connection with another party while receiving another call (Call Waiting), while initiating another call (Call Transfer), or while performing some other call management function. Invoked by keypad sequence.

Call Waiting. Call Waiting notifies an engaged caller of an incoming call and allows them to receive a call from a third party while the party with whom they have been speaking is put on hold. Invoked by keypad sequence.

Call Name Identification. When enabled for a given voip unit (the 'home' voip), this feature gives notice to remote voips involved in calls. Notification goes to the remote voip administrator, not to individual phone stations. When the home voip is the caller, a plain English descriptor will be sent to the remote (callee) voip identifying the channel over which the call is being originated (for example, “Calling Party - Omaha Sales Office Line 2”). If that voip channel is dedicated to a certain individual, the descriptor could say that, as well (for example “Calling Party - Harold Smith in Omaha”). When the home voip receives a call from any remote voip, the home voip sends a status message back to that caller. This message confirms that the home voip’s phone channel is either busy or ringing or that a connection has been made (for example, “Busy Party - Omaha Sales Office Line 2”). These messages appear in the **Statistics – Call Progress** screen of the remote voip.

Note that Supplementary Services parameters are applied on a channel-by-channel basis. However, once you have established a set of supplementary parameters for a particular channel, you can apply this entire set of parameters to another channel by using the **Copy Channel** button and its dialog box. To copy a set of Supplementary Services parameters to all channels, select “Copy to All” and click **Copy**.



The **Supplementary Services** fields are described in the tables below.

Supplementary Services Parameter Definitions		
Field Name	**Values**	**Description**
Select Channel	1-24 (2410); 1-30 (3010)	The channel to be configured is selected here.
Call Transfer Enable	Y/N	Select to enable the Call Transfer function in the voip unit. This is a “blind” transfer and the sequence of events is as follows: Callers A and B are having a conversation. Caller A wants to put B into contact with C. Caller A dials call transfer sequence. Caller A hears dial tone and dials number for caller C. Caller A gets disconnected while Caller B gets connected to caller C. A brief musical jingle is played for the caller on hold.
Transfer Sequence	any phone keypad character	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call transfer. The call-transfer sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #). The sequences for call transfer, call hold, and call waiting can be from 1 to 4 digits in length consisting of any combination of digits 1234567890*#.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Hold Enable	Y/N	Select to enable Call Hold function in voip unit. Call Hold allows one party to maintain an idle (non-talking) connection with another party while receiving another call (Call Waiting), while initiating another call (Call Transfer), or while performing some other call management function.
Hold Sequence	phone keypad characters	The numbers and/or symbols that the caller must press on the phone keypad to initiate a call hold. The call-hold sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #).
Call Waiting Enable	Y/N	Select to enable Call Waiting function in voip unit.
Retrieve Sequence	phone keypad characters, two characters in length	The numbers and/or symbols that the caller must press on the phone keypad to initiate retrieval of a waiting call. The call-waiting retrieval sequence can be 1 to 4 characters in length using any combination of digits or characters (* or #). This is the phone keypad sequence that a user must press to retrieve a waiting call. Customize-able. Sequence should be distinct from sequence that might be used to retrieve a waiting call via the PBX or PSTN.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Name Identification Enable		Enables CNI function. Call Name Identification is not the same as Caller ID. When enabled on a given voip unit currently being controlled by the MultiVOIP GUI (the 'home voip'), Call Name Identification sends an identifier and status information to the administrator of the remote voip involved in the call. The feature operates on a channel-by-channel basis (each channel can have a separate identifier). If the home voip is originating the call, only the **Calling Party** field is applicable. If the home voip is receiving the call, then the **Alerting Party**, **Busy Party**, and **Connected Party** fields are the only applicable fields (and any or all of these could be enabled for a given voip channel). The status information confirms back to the originator that the callee (the home voip) is either busy, or ringing, or that the intended call has been completed and is currently connected. The identifier and status information are made available to the remote voip unit and appear in the **Caller ID** field of its **Statistics – Call Progress** screen. (This is how MultiVOIP units handle CNI messages; in other voip brands, H.450 may be implemented differently and then the message presentation may vary.)

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Calling Party, Allowed Name Type (CNI)		If the 'home' voip unit is originating the call and **Calling Party** is selected, then the identifier (from the **Caller Id** field) will be sent to the remote voip unit being called. The **Caller Id** field gives the remote voip administrator a plain-language identifier of the party that is originating the call occurring on a specific channel. This field is applicable only when the 'home' voip unit is originating the call. **Example**. Suppose a voip system has offices in both Denver and Omaha. In the Omaha voip unit (the 'home' voip in this example), **Call Name Identification** has been enabled, **Calling Party** has been enabled as an **Allowed Name Type,** and "Omaha Sales Office Voipchannel 2" has been entered in the **Caller Id** field. When channel 2 of the Omaha voip is used to make a call to any other voip phone station (for example, the Denver office), the message "Calling Party - Omaha Sales Office Voipchannel 2" will appear in the "Caller Id" field of the **Statistics - Call Progress** screen of the Denver voip.

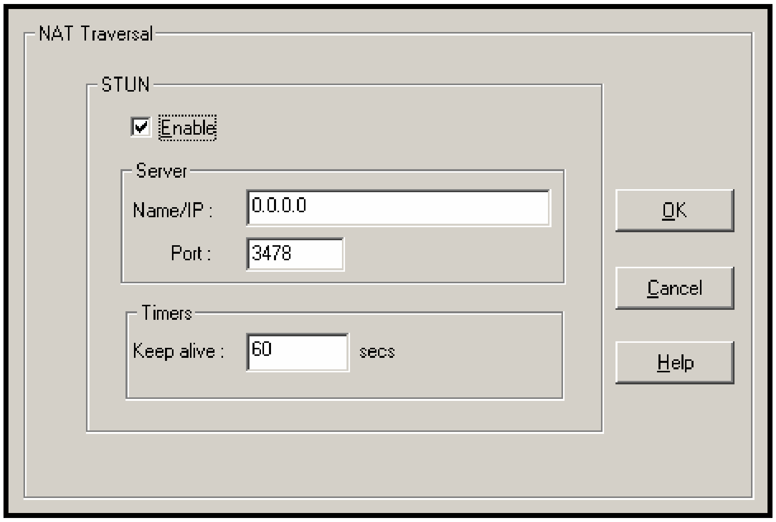
Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Alerting Party, Allowed Name Type (CNI)		If the 'home' voip unit is receiving the call and **Alerting Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote voip unit that the call is ringing. This field is applicable only when the 'home' voip unit is receiving the call. **Example**. Suppose a voip system has offices in both Denver and Omaha. In the Omaha voip unit (the 'home' voip unit in this example), **Call Name Identification** has been enabled, **Alerting Party** has been enabled as an **Allowed Name Type**, and "Omaha Sales Office Voipchannel 2" has been entered in the **Caller Id** field of the **Supplementary Services** screen. When channel 2 of the Omaha voip receives a call from any other voip phone station (for example, the Denver office), the message "Alerting Party - Omaha Sales Office Voipchannel 2" will be sent back and will appear in the **Caller Id** field of the **Statistics – Call Progress** screen of the Denver voip. This confirms to the Denver voip that the phone is ringing in Omaha.

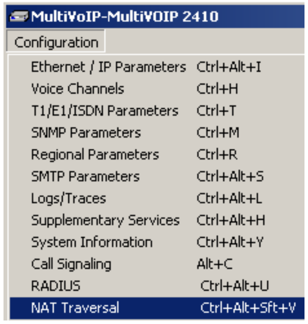
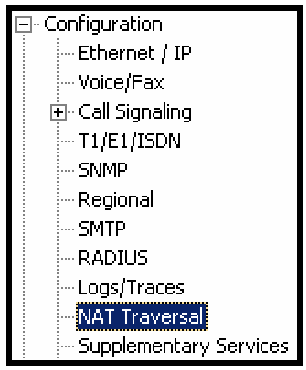
Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Busy Party, Allowed Name Type (CNI)		If the 'home' voip unit is receiving a call directed toward an already engaged channel or phone station and **Busy Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote voip unit that the channel or called party is busy. This field is applicable only when the 'home' voip unit is receiving the call. **Example**. Suppose a voip system has offices in both Denver and Omaha. In the Omaha voip unit (the 'home' voip unit in this example), **Call Name Identification** has been enabled, **Busy Party** has been enabled as an **Allowed Name Type**, and "Omaha Sales Office Voipchannel 2" has been entered in the **Caller Id** field of the **Supplementary Services** screen. When channel 2 of the Omaha voip is busy but still receives a call attempt from any other voip phone station (for example, the Denver office), the message "Busy Party - Omaha Sales Office Voipchannel 2" will be sent back and will appear in the **Caller Id** field of the **Statistics – Call Progress** screen of the Denver voip. This confirms to the Denver voip that the channel or phone station is busy in Omaha.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Connected Party, Allowed Name Type (CNI)		If the 'home' voip unit is receiving a call and **Connected Party** is selected, then the identifier (from the **Caller Id** field) will tell the originating remote voip unit that the attempted call has been completed and the connection is made. This field is applicable only when the 'home' voip unit is receiving the call. **Example**. Suppose a voip system has offices in both Denver and Omaha. In the Omaha voip unit (the 'home' voip unit in this example), **Call Name Identification** has been enabled, **Connected Party** has been enabled as an **Allowed Name Type**, and "Omaha Sales Office Voipchannel 2" has been entered in the **Caller Id** field of the **Supplementary Services** screen. When channel 2 of the Omaha voip completes an attempted call from any other voip phone station (for example, the Denver office), the message "Connect Party - Omaha Sales Office Voipchannel 2" will be sent back and will appear in the **Caller Id** field of the **Statistics – Call Progress** screen of the Denver voip. This confirms to the Denver voip that the call has been completed to Omaha.

Supplementary Services Definitions (cont'd)		
Field Name	**Values**	**Description**
Caller ID		This is the identifier of a specific channel of the 'home' voip unit. The Caller Id field typically describes a person, office, or location, for example, "Harry Smith," or "Bursar's Office," or "Barnesville Factory."
Default	--	When this button is clicked, all Supplementary Service parameters are set to their default values.
Copy Channel	--	Copies the Supplementary Service attributes of one channel to another channel. Attributes can be copied to multiple channels or all channels at once.

17. **Set NAT Traversal** parameters. NAT (Network Address Translation) parameters are applicable only when the MultiVOIP is operating in SIP mode. The use of STUN (Simple Traversal of UDP NATs) servers to aid networks with NAT devices is described in RFC 3489.



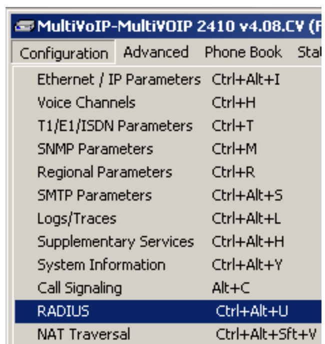
Accessing "NAT Traversal" Parameters	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
Ctrl + Alt + Sft + VH	

Descriptions for NAT Traversal screen fields are presented in the table below.

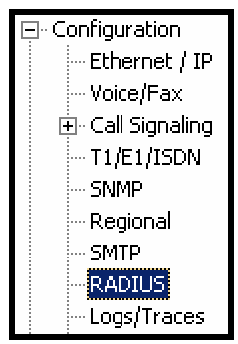
NAT Traversal Definitions (cont'd)		
Field Name	**Values**	**Description**
Enable (STUN)	Y/N	Enables STUN client functionality in the MultiVOIP. STUN (Simple Traversal of UDP through NATs (Network Address Translation)) is a protocol that allows a server to assist client gateways behind a NAT firewall or router with their packet routing.
Name/IP (Server)	n.n.n.n 0 - 255	IP address of the STUN server.
Port (Server; NAT/STUN)	numeric; default= 3478	The data port (TDM time slot) at which STUN info will be transmitted and received.
Keep Alive (Timers; NAT/STUN)	60 – 3600 (in seconds)	The interval at which the STUN client sends indicator ("Keep Alive") packets to the STUN server to determine whether or not the STUN server is available.

18. **Set RADIUS parameters**. In general, RADIUS is concerned with authentication, authorization, and accounting. The MultiVOIP supports the accounting and authentication functions. The accounting function is sell suited for billing of voip telephony services. In the **Attributes** secondary screen (accessed by clicking on Select Attributes), the voip administrator can select the parameters to be tallied by the RADIUS server.

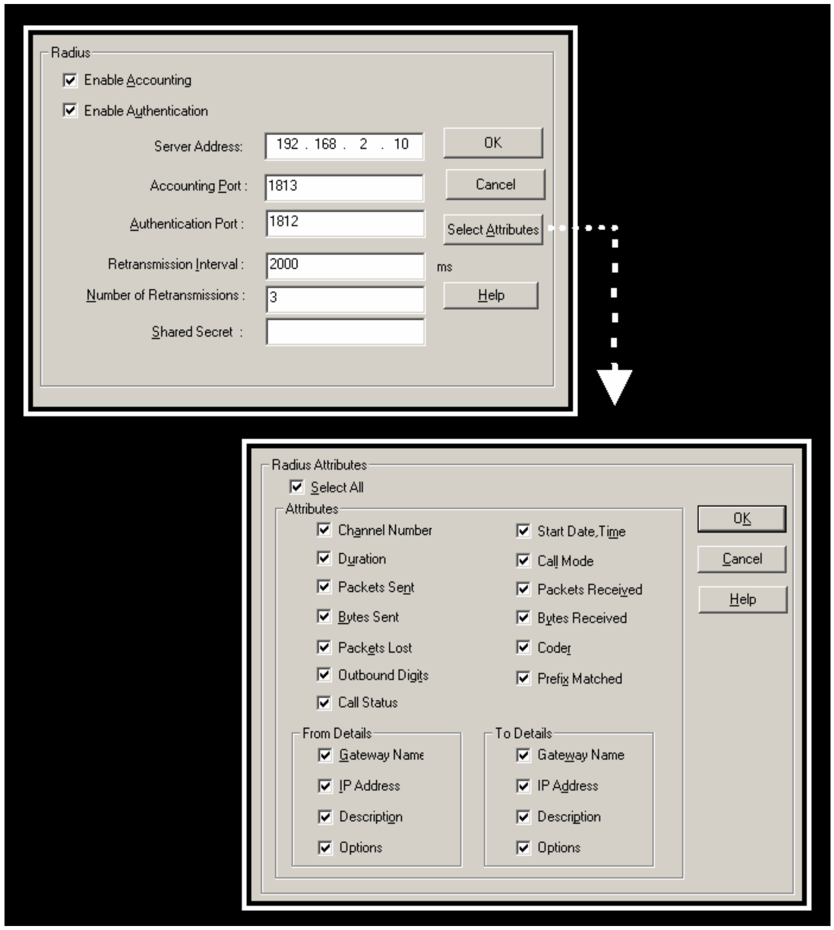
<table>
<tr><th colspan="2">Accessing "RADIUS" Parameters</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>



</td><td>--</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + Alt + U</td><td>



</td></tr>
</table>



The fields of the RADIUS screen are described in the table below.

<table>
<tr><th colspan="3">RADIUS Screen Field Definitions</th></tr>
<tr><th>Field Name</th><th>Values</th><th>Description</th></tr>
<tr><td>Enable Accounting</td><td>Y/N</td><td>When checked, the MultiVOIP will access the accounting functionality of the</td></tr>
<tr><td>Server Address</td><td>n.n.n.n
0 – 255</td><td>IP address of the RADIUS server that handles accounting (billing) for the current MultiVOIP unit.</td></tr>
<tr><td>Accounting Port</td><td>numeric;
1 - 65535</td><td>TDM time slot at which RADIUS accounting information will be transmitted and received.</td></tr>
<tr><td>Retrans-mission Interval</td><td></td><td rowspan="2">If the MultiVOIP sends out a packet to the RADIUS server and doesn't receive a response in the retransmit interval, it will retransmit that packet again and wait the retransmit interval again for a response. How many times it does this is determined by the setting in the Number of Retransmissions field.</td></tr>
<tr><td>Number of Re-transmis-sions</td><td>0 - 255</td></tr>
<tr><td>Shared Secret</td><td>alpha-numeric</td><td>Client encryption key for the current voip unit.</td></tr>
<tr><td>Select Attributes (button)</td><td>--</td><td>Gives access to RADIUS Attributes screen. On Attributes screen, one can specify the parameters to be tallied by the RADIUS server for accounting (usually billing) purposes.</td></tr>
</table>

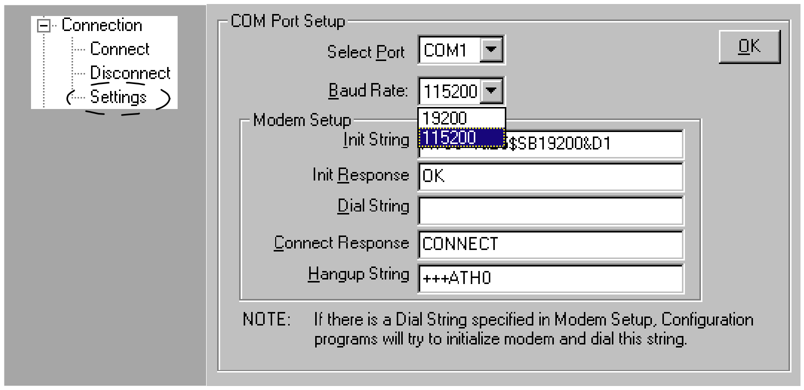
The **RADIUS Parameters** dialog box has a secondary dialog box, **Custom Fields**, that allows you to customize accounting information sent to the RADIUS server by the MultiVOIP. The MultiVOIP software logs data about many aspects of the call traffic going through the MultiVOIP. The Custom Fields screen lets you pick which aspects will be included in the accounting reports sent to the RADIUS server.

"Custom Fields" Definitions			
Field	**Description**	**Field**	**Description**
Select All	Log report to include all fields shown.		
Channel Number	Data channel carrying call.	Start Date, Time	Date and time the phone call began.
Duration	Length of call.	Call Mode	Voice or fax.
Packets Sent	Total packets sent in call.	Packets Received	Total packets received in call.

“Custom Fields” Definitions (cont’d)			
Field	**Description**	**Field**	**Description**
Bytes Sent	Total bytes sent in call.	Bytes Received	Total bytes received in call.
Packets Lost	Packets lost in call.	Coder	Voice Coder /Compression Rate used for call will be listed in log.
Outbound Digits Sent	The DTMF dialing digits received by this gateway from the remote gateway presuming that DTMF is set to "Out of Band."	Prefix Matched	When selected, the phonebook prefix matched in processing the call will be listed in log.
Call Status	Successful or unsuccessful.		
Server Details	The IP address (etc.) of the traffic control server (if any) being used (whether an H.323 gatekeeper, a SIP proxy, or an SPP registrar gateway) will be displayed here if the call is handled through that server. The Options field refers to non-mandatory server features that might be activated. For example, with H.323, various H.323 Version 4 options might be listed (Multiplexing, Tunneling, etc.).		

"Custom Fields" Definitions (cont'd)			
Field	**Description**	**Field**	**Description**
From Details		**To Details**	
Gateway Number	Originating gateway	Gatew N.	Completing or answering gateway
IP Addr	IP address where call originated.	IP Addr	IP address where call was completed or answered.
Descript	Identifier of site where call originated.	Descript	Identifier of site where call was completed or answered.
Options	When selected, log will not use Silence Compression and Forward Error Correction by call originator.	Options	When selected, log will not use Silence Compression and Forward Error Correction by party answering call.

19. **Set Baud Rate**. The **Connection** option in the sidebar menu has a "Settings" item that includes the baud-rate setting for the COM port of the computer running the MultiVOIP software.

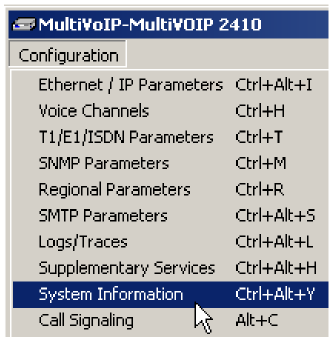
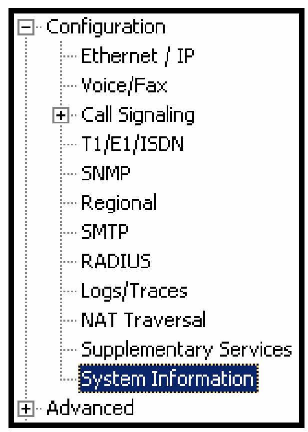


First, it is important to note that the default COM port established by the MultiVOIP program is COM1. ***Do not accept the default value until you have checked the COM port allocation on your PC***. To do this, check for COM port assignments in the system resource dialog box(es) of your Windows operating system. If COM1 is not available, you must change the COM port setting to COM2 or some other COM port that you have confirmed as being available on your PC.

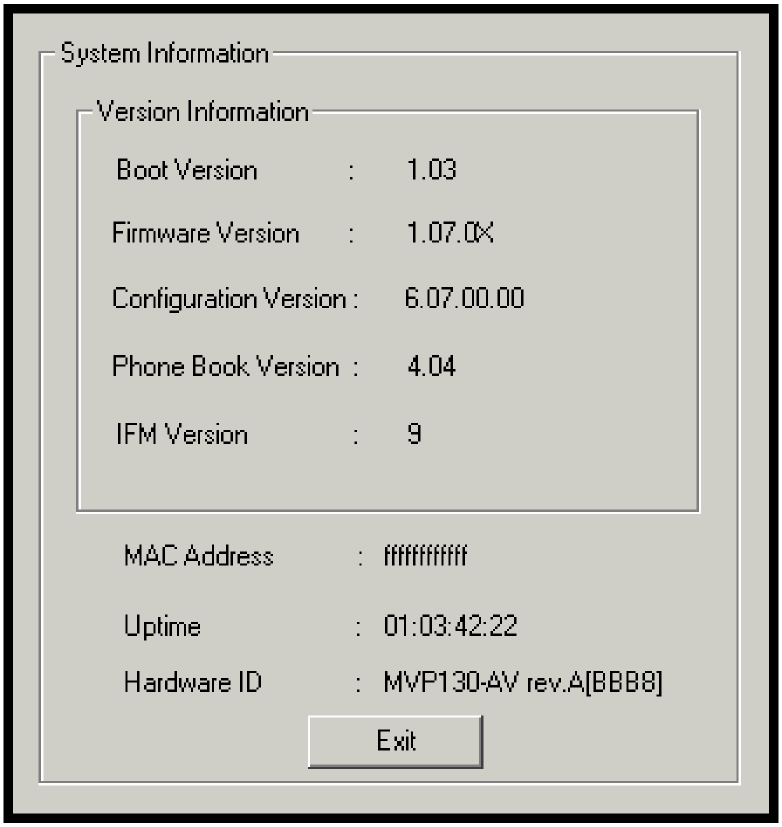
The default baud rate is 115,200 bps.

20. View **System Information** screen and set updating interval (optional).

This dialog box can be reached by pulldown menu, keyboard shortcut, or sidebar.

<table>
<tr><th colspan="2">Accessing “System Information” Screen</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td></td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td>Ctrl + Alt +Y</td><td>
</td></tr>
</table>

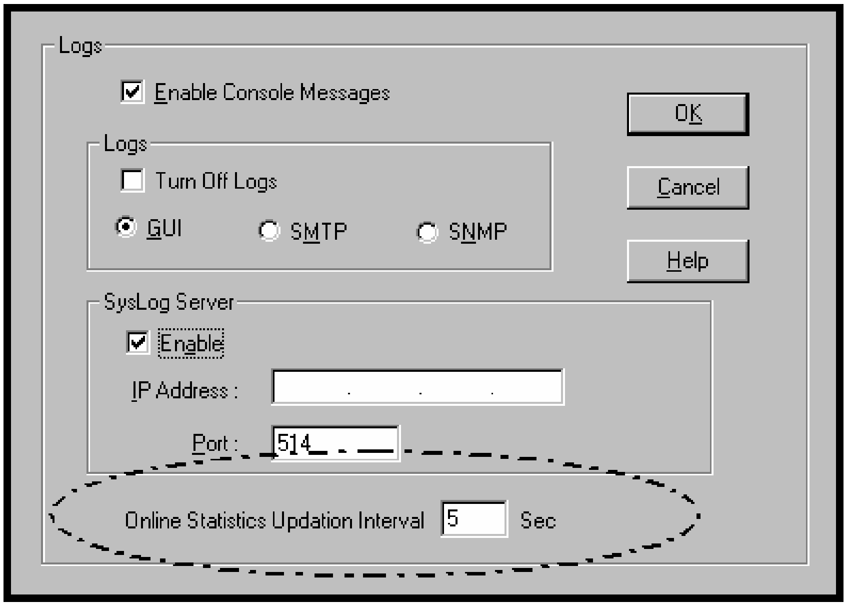
This screen presents vital system information at a glance. Its primary use is in troubleshooting.



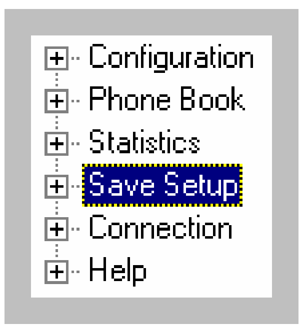
System Information Parameter Definitions		
Field Name	**Values**	**Description**
Boot Version	nn.nn	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Firmware Version	alpha-numeric	Indicates version of MultiVOIP firmware.

System Information Parameter Definitions (cont'd)		
Field Name	**Values**	**Description**
Configur-ation Version	nn.nn.nn.nn alpha-numeric	Indicates version of MultiVOIP Configuration software (which includes screens for IP Parameters, SNMP Parameters, SMTP Parameters, Regional Parameters, etc.
Phone Book Version	numeric	Indicates the version of the inbound and outbound phonebook portion of the MultiVOIP software.
IFM Version	numeric	Indicates the version of the firmware running on the MultiVOIP's Interface Module, which is its analog telephony hardware.
Mac Address	alpha-numeric	Denotes the number assigned as the voip unit's unique Ethernet address.
Up Time	days: hours: mm:ss	Indicates how long the voip has been running since its last booting.
Hardware ID	alpha-numeric	Indicates the version of the MultiVOIP unit's circuit board and components.

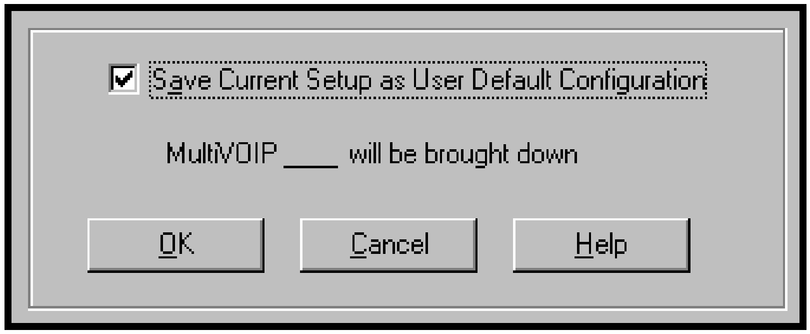
The frequency with which the System Information screen is updated is determined by a setting in the Logs screen



21. **Saving the MultiVOIP Configuration**. When values have been set for all of the MultiVOIP's various operating parameters, click on **Save Setup** in the sidebar.



22. **Creating a User Default Configuration**. When a "Setup" (complete grouping of parameters) is being saved, you will be prompted about designating that setup as a "User Default" setup. A User Default setup may be useful as a baseline of site-specific values to which you can easily revert. Establishing a User Default Setup is optional.



Chapter 6: T1 Phonebook Configuration

(North American Telephony Standards)

T1 Versus E1 Telephony Environments

We present separate chapters for the MVP2410 MultiVOIP (this chapter) and the MVP3010 MultiVOIP (Chapter 7) because the respective telephony environments in which they operate have different standards and conventions. The MVP2410 is designed to operate under North American or T1 standards; the MVP3010 is designed to operate under European or E1 standards. The configuration of the phonebook is the same in either case. However, differences in the telephony environment give rise to different examples in each case. Series II analog MultiVOIP units (MVP130, MVP130FXS, MVP210, MVP410, and MVP810) can be operated in either the T1 or E1 environments. The examples in this chapter show these analog voip units being used in the same system as the MVP2410 digital MultiVOIP.

Configuring T1 (NAM) Telephony MultiVOIP Phonebooks

When a VoIP serves a PBX system, it's important that the operation of the VoIP be transparent to the telephone end user. That is, the VoIP should not entail the dialing of extra digits to reach users elsewhere on the network that the VoIP serves. On the contrary, VOIP service more commonly reduces dialed digits by allowing users (served by PBXs in facilities in distant cities) to dial their co-workers with 3-, 4-, or 5-digit extensions as if they were in the same facility.

Furthermore, the setup of the VoIP generally should allow users to make calls on a non-toll basis to any numbers accessible without toll by users at all other locations on the VoIP system. Consider, for example, a company with VOIP-equipped offices in New York, Miami, and Los Angeles, each served by its own PBX. When the VOIP phone books are set correctly, personnel in the Miami office should be able to make calls without toll not only to the company's offices in New York and Los Angeles, but also to any number that's local in those two cities.

To achieve transparency of the VoIP telephony system and to give full access to all types of non-toll calls made possible by the VOIP system, the VoIP administrator must properly configure the "Outbound" and "Inbound" phone-books of each VoIP in the system.

The "Outbound" phonebook for a particular VoIP unit describes the dialing sequences required for a call to originate locally (typically in a PBX in a particular facility) and reach any of its possible destinations at remote VoIP sites, including non-toll calls completed in the PSTN at the remote site.

The "Inbound" phonebook for a particular VoIP unit describes the dialing sequences required for a call to originate remotely from any other VOIP sites in the system, and to terminate on that particular VOIP.

Briefly stated, *the MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook describes the dialing sequences that can be used to call that MultiVOIP and how those calls will be directed.* (Of course, the phone numbers are not literally "listed" individually, but are, instead, described by rule.)

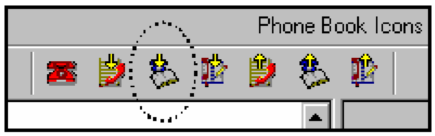
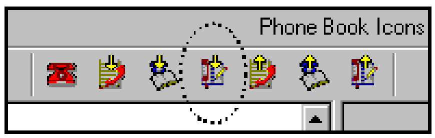
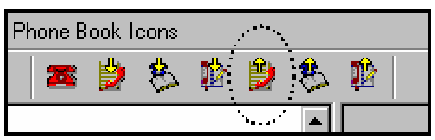
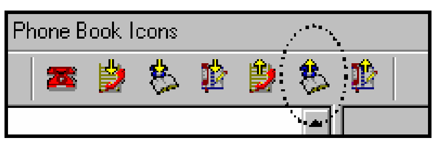
Consider two types of calls in the three-city system described above: (1) calls originating from the Miami office and terminating in the New York (Manhattan) office, and (2) calls originating from the Miami office and terminating in New York City but off the company's premises in an adjacent area code, an area code different than the company's office but still a local call from that office (e.g., Staten Island).

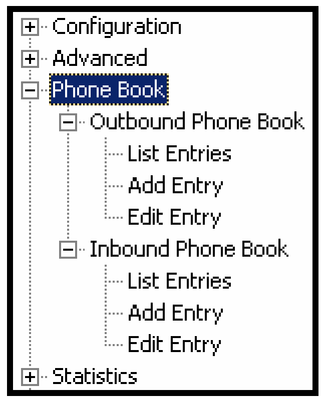
The first type of call requires an entry in the Outbound PhoneBook of the Miami VOIP and a coordinated entry in the Inbound phonebook of the New York VOIP. These entries would allow the Miami caller to dial the New York office as if its phones were extensions on the Miami PBX.

The second type of call similarly requires an entry in the Outbound PhoneBook of the Miami VOIP and a coordinated entry in the Inbound Phonebook of the New York VOIP. However, these entries will be longer and more complicated. Any Miami call to New York City local numbers will be sent through the VOIP system rather than through the regular toll public phone system (PSTN). But the phonebook entries can be arranged so that the VOIP system is transparent to the Miami user, such that even though that Miami user dials the New York City local number just as they would through the public phone system, that call will still be completed through the VOIP system.

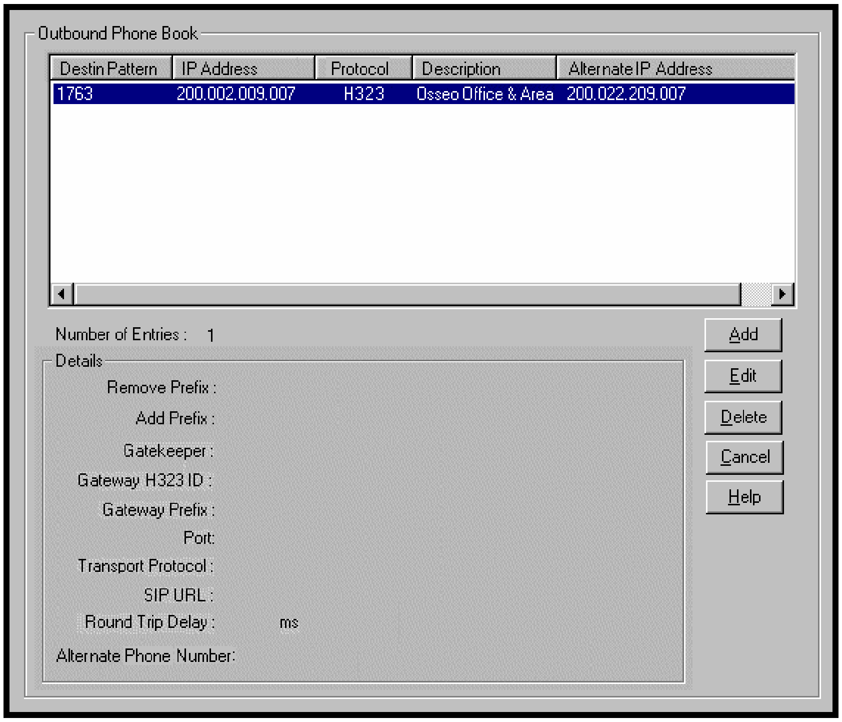
This PhoneBook Configuration procedure is brief, but it is followed by an example case. For many people, the example case may be easier to grasp than the procedure steps. Configuration is not difficult, but all phone number sequences and other information must be entered exactly; otherwise connections will not be made.

Phonebook configuration screens can be accessed using icons or the sidebar menu.

Phonebook Icons	Description
	Phonebook Configuration
	Inbound Phonebook Entries List
	Add Inbound Phonebook Entry
	Edit selected Inbound Phonebook Entry
	Outbound Phonebook Entries List
	Add Outbound Phonebook Entry
	Edit selected Outbound Phonebook Entry

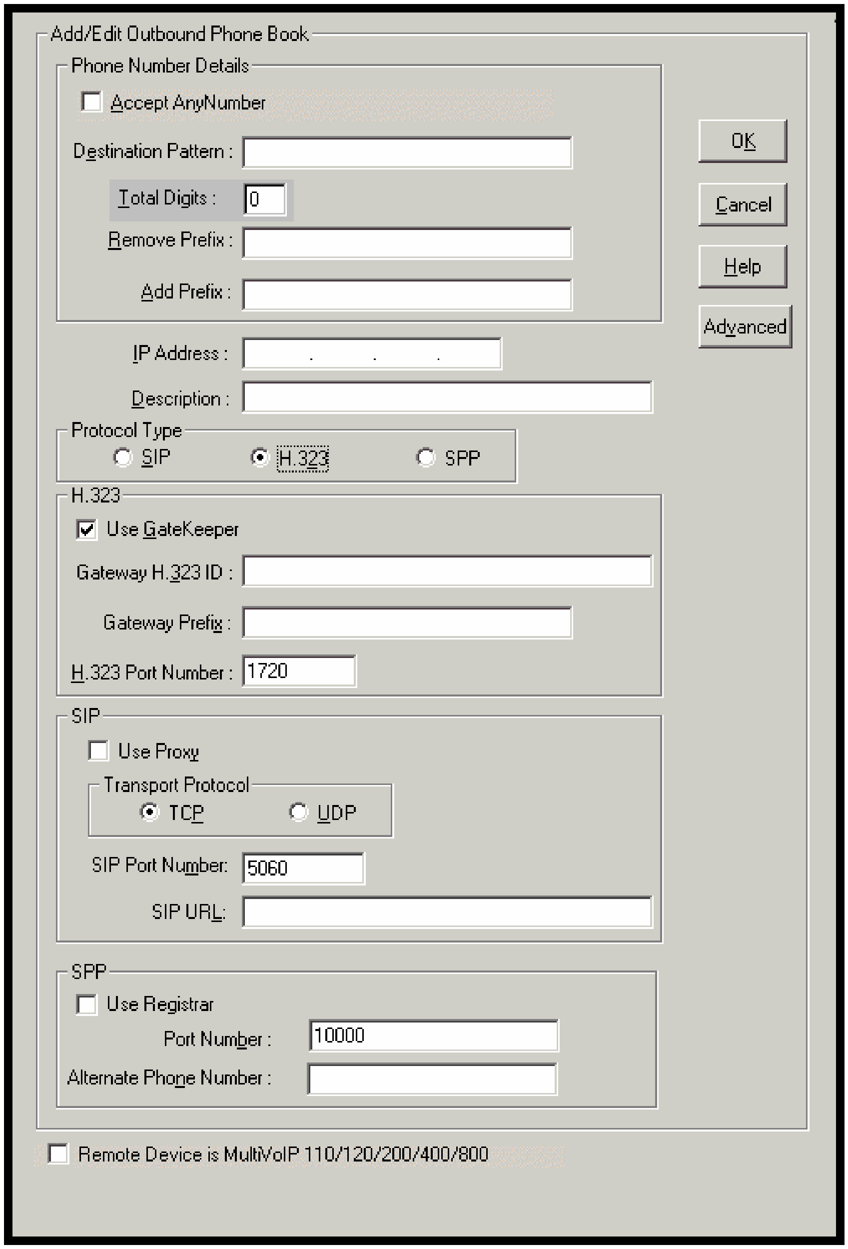
Phonebook Pulldown Menu	
	
Inbound Phonebook Shortcut	**Outbound Phonebook Shortcut**
Alt + I	Alt + O
Phonebook Sidebar Menu	
	

1. Select **Outbound Phone Book/List Entries**.
   Fields in the "Details" section will differ depending on the protocol (H.323, SIP, or SPP) of the selected list entry to which the details pertain.



Click **Add**.

2. The **Add/Edit Outbound PhoneBook** screen appears.



Enter Outbound PhoneBook data for your MultiVOIP unit. Note that the Advanced button gives access to the Alternate IP Routing feature, if needed. Alternate IP Routing can be implemented in a secondary screen (as described after the primary screen field definitions below).

The fields of the **Add/Edit Outbound Phone Book** screen are described in the table below.

Add/Edit Outbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Accept Any Number	Y/N	When checked, "Any Number" appears as the value in the Destination Pattern field. The **Any Number** feature works differently depending on whether or not an external routing device is used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Any Number** is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the **IP Address** in the Add/Edit Outbound Phone Book screen. "Any Number" can be used in addition to one or more Destination Patterns. **When external routing device is used**. If Any Number is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the external routing device used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). The IP Address of the external routing device must be set in the Phone Book Configuration screen.

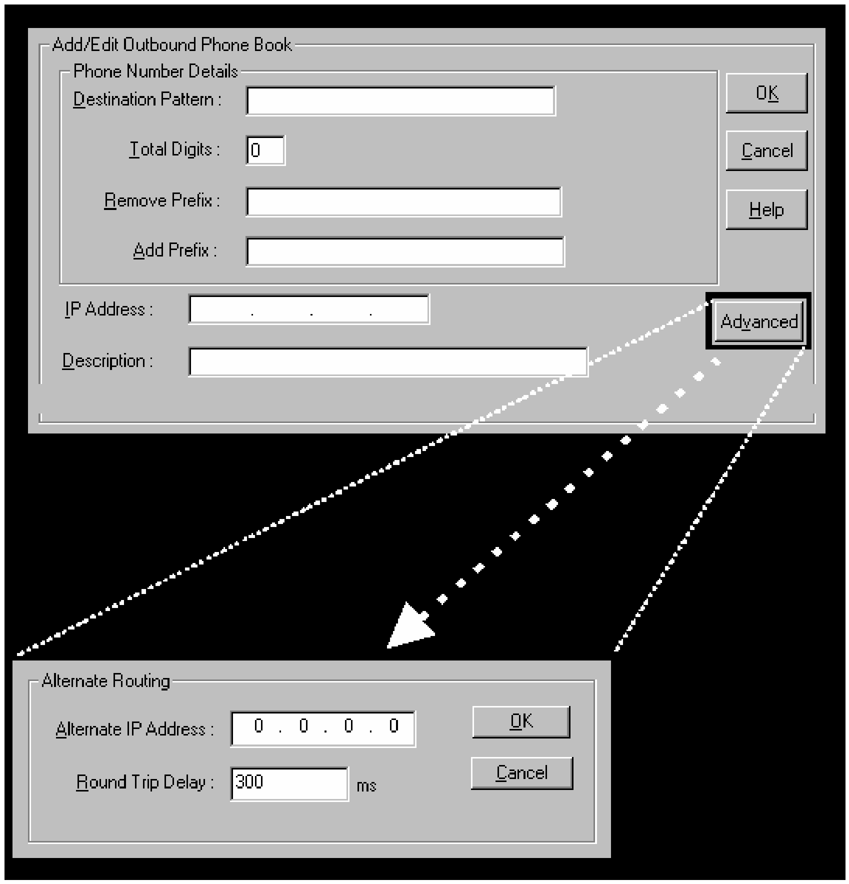
Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be connected to another VOIP in the system. Numbers beginning with these sequences are diverted from the PTSN and carried on Internet or other IP network.
Total Digits	as needed	*This field currently disabled.* number of digits the phone user must dial to reach specified destination.
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination
Add Prefix	dialed digits	digits to be added before completing call to destination
IP Address	n.n.n.n for n = 0-255	the IP address to which the call will be directed if it begins with the destination pattern given
Description	alpha-numeric	Describes the facility or geographical location at which the call will be completed.
Protocol Type	SIP or H.323 or SPP	Indicates protocol to be used in outbound transmission. Single Port Protocol (SPP) is a non-standard protocol designed by Multi-Tech.

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 fields		
Use Gatekeepr	Y/N	Indicates whether or not gatekeeper is used.
Gateway H.323 ID	alpha-numeric	The H.323 ID assigned to the destination MultiVOIP. Only valid if "Use Gatekeeper" is enabled for this entry.
Gateway Prefix	numeric	This number becomes registered with the GateKeeper. Call requests sent to the gatekeeper and preceded by this prefix will be routed to the VOIP gateway.
H.323 Port Number	1720	This parameter pertains to Q.931, which is the H.323 call signaling protocol for setup and termination of calls (aka ITU-T Recommendation I.451). H.323 employs only one "well-known" port (1720) for Q.931 signaling. If Q.931 message-oriented signaling protocol is used, 1720 must be chosen as the H.323 Port Number.

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
SIP Fields		
Use Proxy	Y/N	Select if proxy server is used.
Transport Protocol	TCP or UDP	Voip administrator must choose between UDP and TCP transmission protocols. UDP is a high-speed, low-overhead connectionless protocol where data is transmitted without acknowledgment, guaranteed delivery, or guaranteed packet sequence integrity. TCP is slower connection-oriented protocol with greater overhead, but having acknowledgment and guarantees delivery and packet sequence integrity.
SIP Port Number	5060 or other *See RFC 3087 ("Control of Service Context using SIP Request-URI," by the Network Working Group).	The SIP Port Number is a UDP logical port number. The voip will "listen" for SIP messages at this logical port. If SIP is used, 5060 is the default, standard, or "well known" port number to be used. If 5060 is not used, then the port number used is that specified in the SIP Request URI (Universal Resource Identifier).
SIP URL	*sip.userphone* @ *hostserver*, where "userphone" is the telephone number and "hostserver"is the domain name or an address on the network	Looking similar to an email address, a **SIP URL** identifies a user's address. In SIP communications, each caller or callee is identified by a SIP url: sip:user_name@host_name. The format of a sip url is very similar to an email address, except that the "sip:" prefix is used.

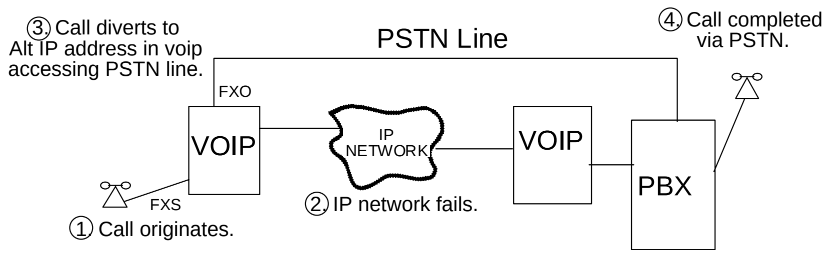
Add/Edit Outbound Phone Book: Field Def'ns (cont'd)		
Field Name	**Values**	**Description**
SPP Fields		
Use Registrar	**Values:** Y/N **Description:** Select this checkbox to use registrar when voip system is operating in the "Registrar/Client" SPP mode. In this mode, one voip (the registrar, as set in Phonebook Configuration screen) has a static IP address and all other voips (clients) point to the registar's IP address as functionally their own. However, if your voip system overall is operating in "Registrar/Client" mode but you want to make an exception and use Direct mode for the destination pattern of this particular Add/Edit Phonebook entry, leave this checkbox unselected. Leave this checkbox unselected if your overall voip system is operating in the "Direct" SPP mode. In this mode, all voips in system are peers and each has its own static IP address.	
Port Number	**Values:** numeric **Description:** When operating in "Registrar/Client" mode, this is the port by which the gateway receives all SPP data and control messages from the registrar gateway. (This ability to receive all data and messages via one port allows the voip to operate behind a firewall with only one port open.) When operating in "Direct" mode, this is the Port by which peer voips receive data and messages.	
Alternate Phone Number	numeric	Phone number associated with alternate IP routing.
Remote Device is [legacy voip]	Y/N	When checked, this MultiVOIP can operate with 'first-generation' MultiVOIP units in the same IP network. These include MVP-110/120/200/400/800.
Advanced button	**Values:** N/A **Description:** Gives access to secondary screen where an Alternate IP Route can be specified for backup or redundancy of signal paths. See discussion on next page. For SIP & H.323 operation only.	

Clicking on the **Advanced** button brings up the **Alternate Routing** secondary screen. This feature provides an alternate path for calls if the primary IP network cannot carry the traffic. Often in cases of failure, call traffic is temporarily diverted into the PSTN. However, this feature could also be used to divert traffic to a redundant (backup) unit in case one voip unit fails. The user must specify the IP address of the alternate route for each destination pattern entry in the Outbound Phonebook.



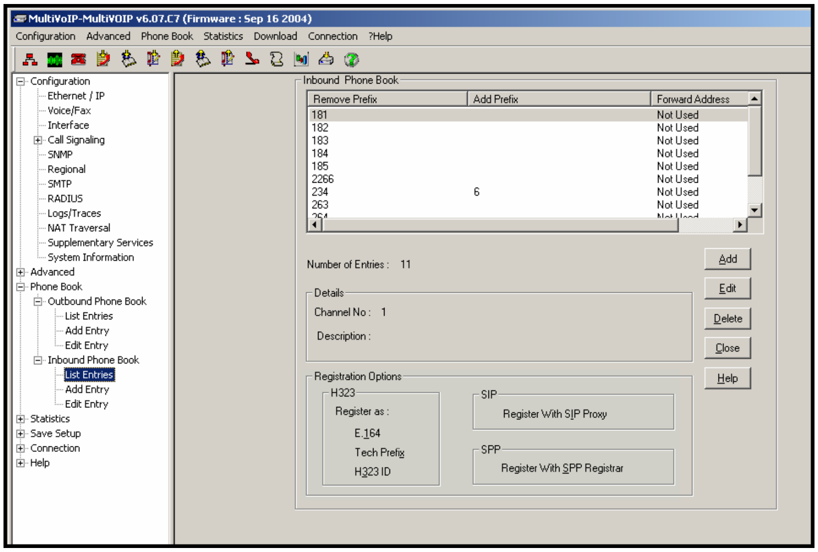
Alternate Routing Field Definitions		
Field Name	**Values**	**Description**
Alternate IP Address	n.n.n.n where n= 0-255	Alternate destination for outbound data traffic in case of excessive delay in data transmission.
Round Trip Delay	milliseconds	The Round Trip Delay is the criterion for judging when a data pathway is considered blocked. When the delay exceeds the threshold specified here, the data stream will be diverted to the alternate destination specified as the Alternate IP Address.

The Alternate Routing function facilitates PSTN Failover protection, that is, it allows you to re-route voip calls automatically over the PSTN if the voip system fails. The MultiVOIP can be programmed to respond to excessive delays in the transmission of voice packets, which the MultiVOIP interprets as a failure of the IP network. Upon detecting an excessive delay in transmission of voice packets (overly high "latency" in the network) the MultiVOIP diverts the call to another IP address, which itself is connected to the PSTN (for example, via an FXO port on the self-same MultiVOIP could be connected to the PSTN).

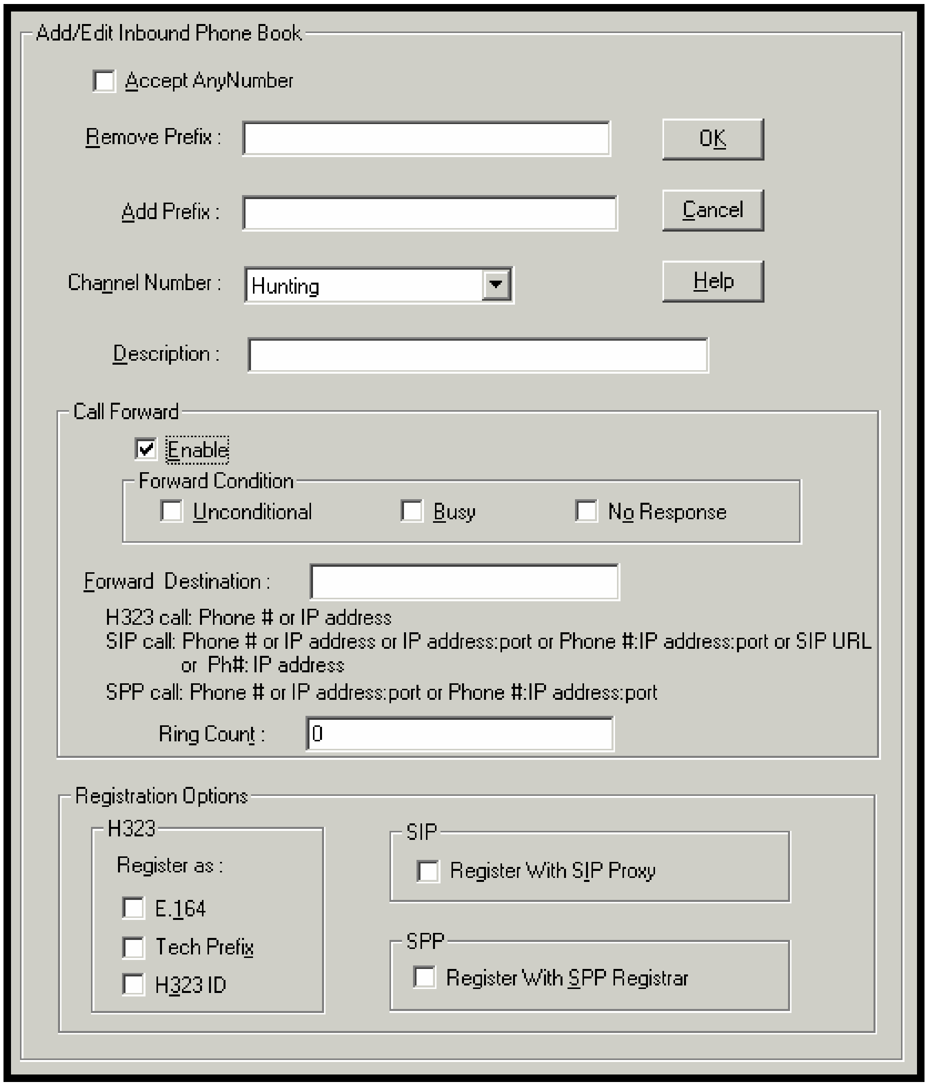


PSTN Failover Feature. The MultiVOIP can be programmed to divert calls to the PSTN temporarily in case the IP network fails.

3. Select **Inbound PhoneBook | List Entries**.



4. The **Add/Edit Inbound PhoneBook** screen appears.



Enter Inbound PhoneBook data for your MultiVOIP. The fields of the Add/Edit Inbound PhoneBook screen are described in the table below.

Add/Edit Inbound Phone Book: Field Definitions

Field Name	Values	Description
Accept Any Number	**Values**: Y/N	**Description:** When checked, "Any Number" appears as the value in the Remove Prefix field. The **Any Number** feature of the Inbound Phone Book does not work when an external routing device is used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Any Number** is selected, calls received from phone numbers not matching a listed Prefix (shown in the Remove Prefix column of the Inbound Phone Book) will be admitted into the voip on the channel listed in the **Channel Number** field. "Any Number" can be used in addition to one or more Prefixes.
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination (often a local PBX)
Add Prefix	dialed digits	digits to be added before completing call to destination (often a local PBX)
Channel Number	1-24, or "Hunting"	T1 channel number to which the call will be assigned as it enters the local telephony equipment (often a local PBX). "Hunting" directs the call to any available channel.
Description	--	Describes the facility or geographical location at which the call originated.
Call Forward Parameters		
Enable	Y/N	Click the check-box to enable the call-forwarding feature.

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Forward Parameters		
Forward Condition	Uncondit.; Busy No Resp.	**Unconditional**. When selected, all calls received will be forwarded. **Busy**. When selected, calls will be forwarded when station is busy. **No Response**. When selected, calls will be forwarded if called party does not answer after a specified number of rings, as specified in **Ring Count** field. Forwarding can be conditioned on both "Busy" and "No Response."
Forward Destination IP address, phone number, port number, etc.	Phone number or IP address to which calls will be directed. For H.323 calls, the Forward Destination can be either a Phone Number or an IP Address. For SIP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address, (c) IP address: port number, (d) phone number:IP addr: port number, (e) SIP URL, or (f) phone #: IP address. For SPP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address: port, or (c) phone number: IP address: port.	

Add/Edit Inbound Phone Book: Field Definitions (cont'd)	
Field Name	**Values and Description**
Ring Count	0, 1, 2, 3, etc. When "No Response" is condition for forwarding calls, this determines how many unanswered rings are needed to trigger the forwarding.
Registration Option Parameters	In an H.323 voip system, gateways can register with the system using one of these identifiers: (a) an E.164 identifier, (b) a Tech Prefix identifier, or (c) an H.323 ID identifier. In a SIP voip system, gateways can register with the SIP Proxy. In an SPP voip system, gateways can register with the SPP Registrar voip unit.

5. When your Outbound and Inbound PhoneBook entries are completed, click on **Save Setup** in the sidebar menu to save your configuration.

You can change your configuration at any time as needed for your system.

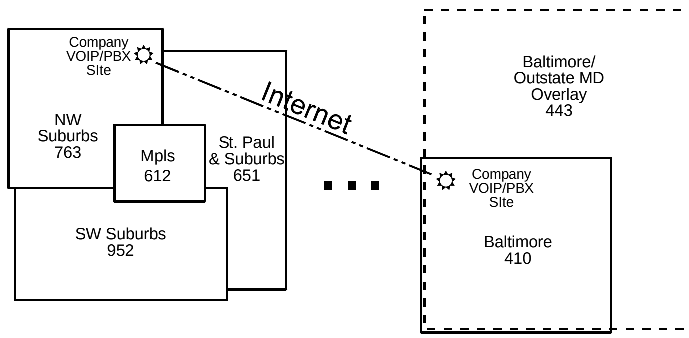
Remember that the initial MultiVOIP setup must be done locally or via the built-in Remote Configuration/Command Modem using the MultiVOIP program. After the initial configuration is complete, all of the MultiVOIP units in the VOIP system can be configured, re-configured, and updated from one location using the MultiVOIP web GUI software program or the MultiVOIP program (in conjunction with the built-in modem).

T1 Phonebook Examples

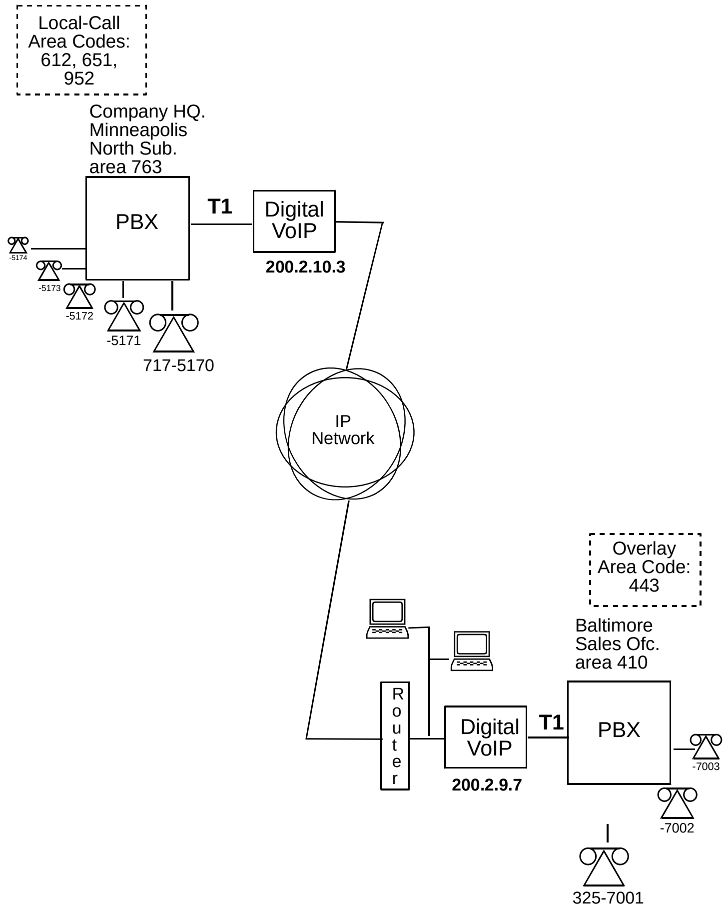
The following example demonstrates how Outbound and Inbound PhoneBook entries work in a situation of multiple area codes. Consider a company with offices in Minneapolis and Baltimore.

3 Sites, All-T1 Example

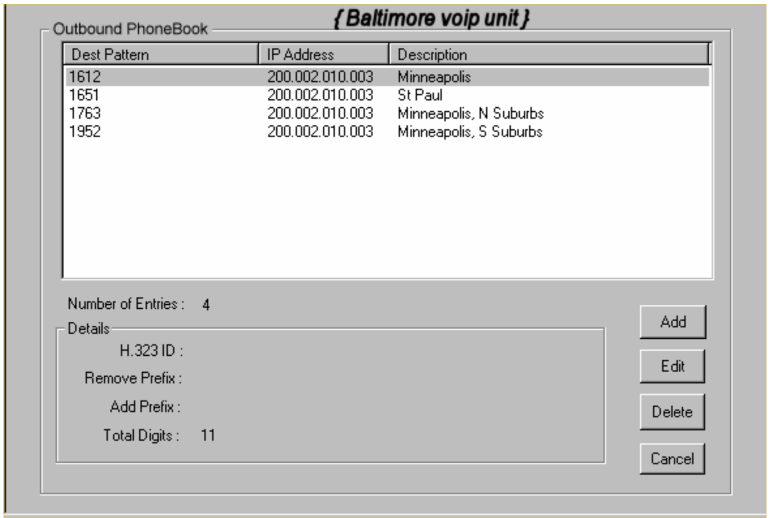
Notice first the area code situation in those two cities: Minneapolis's local calling area consists of multiple adjacent area codes; Baltimore's local calling area consists of a base area code plus an overlay area code.



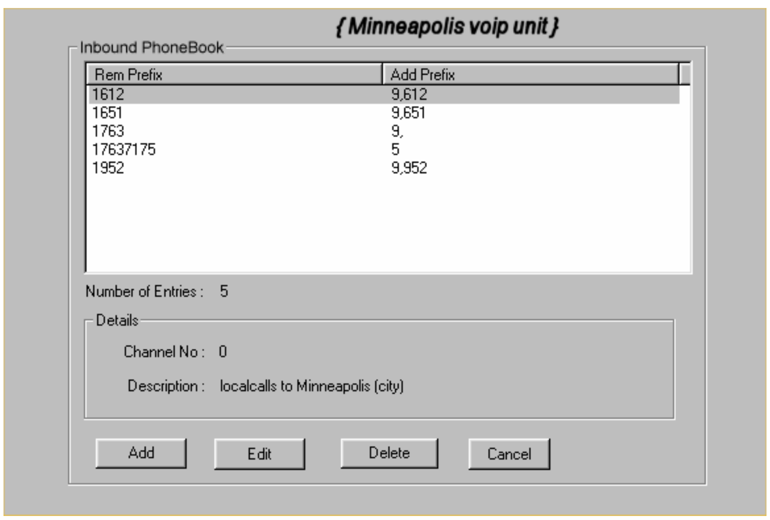
An outline of the equipment setup in both offices is shown below.



The screen below shows Outbound PhoneBook entries for the VOIP located in the company's Baltimore facility.



The entries in the Minneapolis VOIP's Inbound PhoneBook match the Outbound PhoneBook entries of the Baltimore VOIP, as shown below.



To call the Minneapolis/St. Paul area, a Baltimore employee must dial eleven digits. (In this case, we are assuming that the Baltimore PBX does not require an "8" or "9" to seize an outside phone line.)

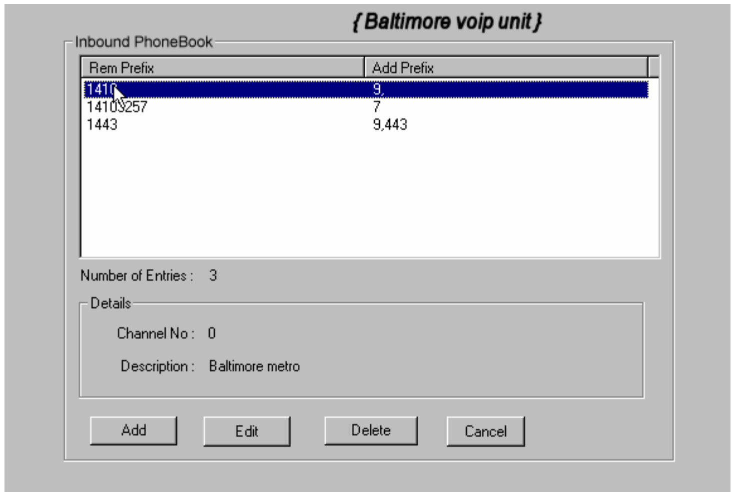
If a Baltimore employee dials any phone number in the 612 area code, the call will automatically be handled by the company's voip system. Upon receiving such a call, the Minneapolis voip will remove the digits "1612". But before the suburban-Minneapolis voip can complete the call to the PSTN of the Minneapolis local calling area, it must dial "9" (to get an outside line from the PBX) and then a comma (which denotes a pause to get a PSTN dial tone) and then the 10-digit phone number which includes the area code (612 for the city of Minneapolis; which is different than the area code of the suburb where the PBX is actually located -- 763).

A similar sequence of events occurs when the Baltimore employee calls number in the 651 and 952 area codes because number in both of these area codes are local calls in the Minneapolis/St. Paul area.

The simplest case is a cal from Baltimore to a phone within the Minneapolis/St. Paul area code where the company's voip and PBX are located, namely 763. In that case, that local voip removes 1763 and dials 9 to direct the call to its local 7-digit PSTN.

Finally, consider the longest entry in the Minneapolis Inbound Phonebook, "17637175. Note that the main phone number of the Minneapolis PBX is 763-717-5170. The destination pattern 17637175 means that all calls to Minneapolis employees will stay within the suburban Minneapolis PBX and will not reach or be carried on the local PSTN.

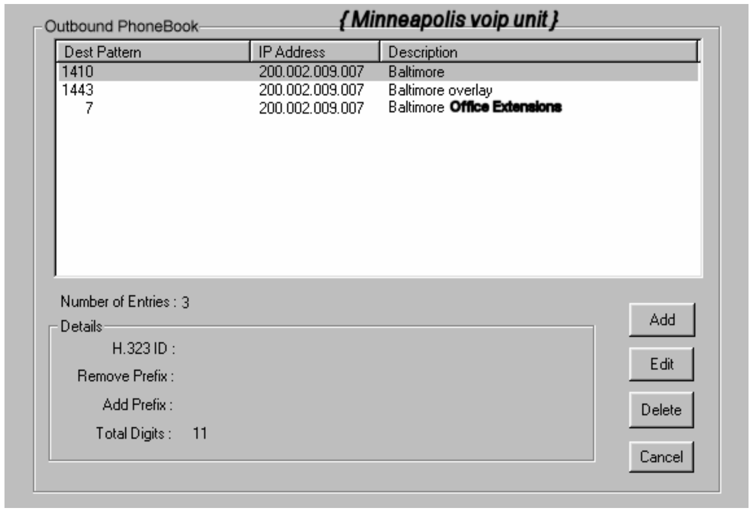
Similarly, the Inbound PhoneBook for the Baltimore VOIP (shown first below) generally matches the Outbound PhoneBook of the Minneapolis VOIP (shown second below).



Notice the extended prefix to be removed: 14103257. This entry allows Minneapolis users to contact Baltimore co-workers as though they were in the Minneapolis facility, using numbers in the range 7000 to 7999.

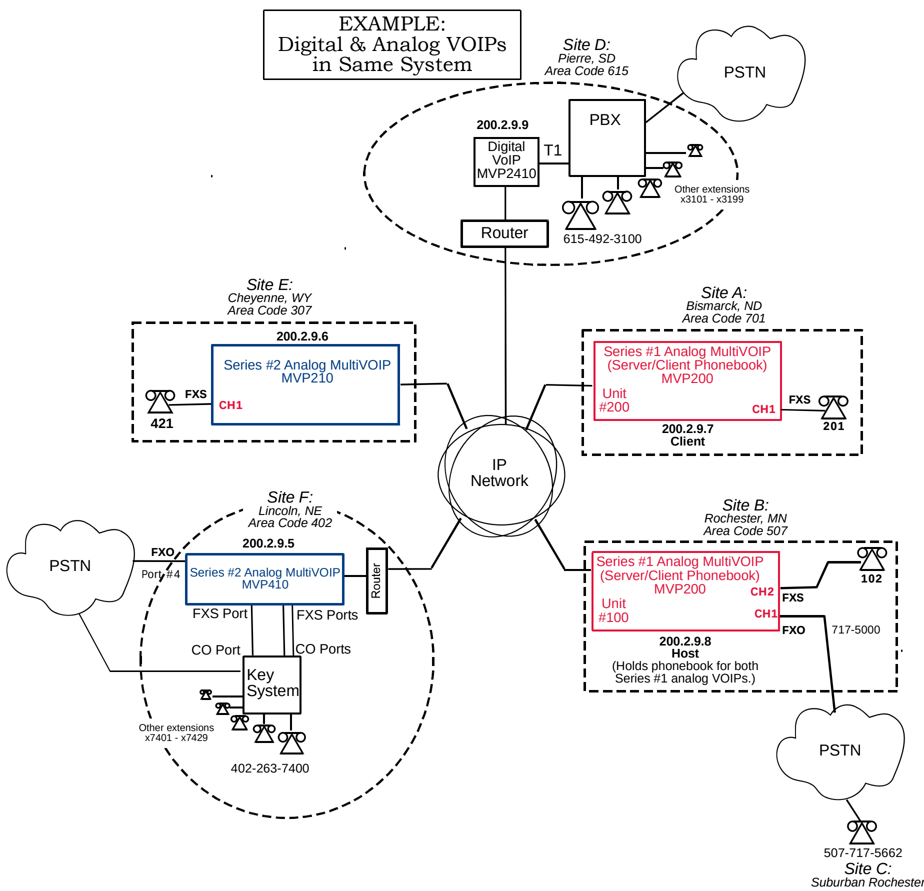
Note also that a comma (as in the entry 9,443) denotes a delay in dialing. A one-second delay is commonly used to allow a second dial tone to be generated for calls going outside of the facility's PBX system.

The Outbound PhoneBook for the Minneapolis VOIP is shown below. The third destination pattern, "7" facilitates reception of co-worker calls using local-appearing-extensions only. In this case, the "Add Prefix" field value for this phonebook entry would be "1410325" .



Configuring Mixed Digital/Analog VOIP Systems

Analog MultiVOIP units, like the MVP-210/410/810 are compatible with digital MultiVOIP units like the MVP2410. In many cases, digital and analog VOIP units will appear in the same telephony/IP system. In addition to MVP-210/410/810 MultiVOIP units (Series II units), legacy analog VOIP units (Series I units made by MultiTech) may be included in the system, as well. When legacy VOIP units are included, the VOIP administrator must handle two styles of phonebooks in the same VOIP network. The diagram below shows a small-scale system of this kind: one digital VOIP (the MVP2410) operates with two Series II analog VOIPs (an MVP210 and an MVP410), and two Series I legacy VOIPs (two MVP200 units).



The Series I analog VOIP phone book resides in the “Host” VOIP unit at Site B. It applies to both of the Series I analog VOIP units.

Each of the Series II analog MultiVOIPs (the MVP210 and the MVP410) requires its own inbound and outbound phonebooks. The MVP2410 digital MultiVOIP requires its own inbound and outbound phonebooks, as well.

These seven phone books are shown below.

Phone Book for Series I Analog VOIP Host Unit (Site B)			
VOIP Dir # -OR- Destination Pattern	IP Address	Channel	Comments
102	200.2.9.8	2	Site B, FXS channel.
101	200.2.9.8	1	Site B, FXO channel.
421	200.2.9.6	0	Site E FXS channel.
201	200.2.9.7	1	Site A, FXS channel.
1615 xxx xxxx	200.2.9.9	0 (Note 2.)	Gives remote voip users access to local PSTN of Site D (Pierre, SD, area code 615).
3xxx (Note 1.)	200.2.9.9	0	Allows remote voip users to call all PBX extensions at Site D (Pierre, SD) using only four digits.
1402	200.2.9.5	0	Gives remote voip users access to local PSTN of Site F (Lincoln, NE; area code 402).
140226374 (Note 1) (Note 3)	200.2.9.5	0	Gives remote voip users access to key phone system extensions at Site F (Lincoln).

Note 1. The “x” is a wildcard character.

Note 2. By specifying “Channel 0,” we instruct the MVP2400/2410 to choose any available data channel to carry the call.

Note 3. Note that Site F key system has only 30 extensions (x7400-7429). This destination pattern (140226374) actually directs calls to 402-263-74**30** through 402-263-74**99** into the key system, as well.
This means that such calls, which belong on the PSTN, cannot be completed. In some cases, this might be inconsequential because an entire exchange (fully used or not) might have been reserved for the company or it might be unnecessary to reach those numbers. However, to specify only the 30 lines actually used by the key system, the destination pattern 140226374 would have to be replaced by three other destination patterns, namely 140226374**0**, 140226374**1**, and 140226374**2**. In this way, calls to 402-263-74**30** through 402-263-74**99** would be properly directed to the PSTN. In the Site D outbound phonebook, the 30 lines are defined exactly, that is, without making any adjacent phone numbers unreachable through the voip system.

Outbound Phone Book for MVP2410 Digital VOIP (Site D)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Bismarck).
1507	1507	101# Note 3.	200.2.9.8	To originate calls to Rochester local PSTN using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP.
421			200.2.9.6	Calls to Site E (Cheyenne).
1402			200.2.9.5	Calls to Lincoln area local PSTN (via FXO channel, CH4, of the Site F VOIP).
1402 263 740			200.2.9.5	Calls to extensions (thirty) of key system at Site F (Lincoln). Human operator or auto-attendant is needed to complete these calls.
1402 263 741			200.2.9.5	
1402 263 742			200.2.9.5	
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP2410 Digital VOIP (Site D)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comment**
1615	9, Note 4. Note 5.	0	Allows phone users at remote voip sites to call non-toll numbers within the Site D area code (615; Pierre, SD) over the VOIP network.
1615 49231	31	0	Allows voip calls directly to employees at Site D (at extensions x3101 to x3199).
Note 4. "9" gives PBX station users access to outside line. Note 5. The comma represents a one-second pause, the time required for the user to receive a dial tone on the outside line (PSTN). The comma is only allowed in the Inbound phonebook.			

Outbound Phone Book for MVP410 Analog VOIP (Site F)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Bismarck).
1507	1507	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Rochester area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Rochester).
421			200.2.9.6	Calls to Site E (Cheyenne).
1615			200.2.9.9	Calls to Pierre area PSTN via Site D PBX.
31		1615 492	200.2.9.9	Calls to Pierre PBX extensions with four digits.
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP410 Analog VOIP (Site F)

Remove Prefix	Add Prefix	Channel Number	Comment
1402		4	Access to Lincoln local PSTN by users at remote VOIP locations via FXO port at Site F.
1402 263740	740	0	Gives remote voip users access to extension of key phone system at Site F (Lincoln). Because call is completed at key system, abbreviated dialing (4 digits) is not workable. Human operator or auto-attendant is needed to complete these calls.
1402 263741	741	0	
1402 263742	742	0	

Outbound Phone Book for MVP210 Analog VOIP (Site E)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A.
1507	1507	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Rochester area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP.
1402			200.2.9.5	Calls to Lincoln area PSTN (via FXO channel, CH4, of the Site F VOIP).
7		1402 263	200.2.9.5	Calls to Lincoln key extensions with four digits.
1615			200.2.9.9	Calls to Pierre area PSTN via Site D PBX.
31		1615 492	200.2.9.9	Calls to Pierre PBX extensions with four digits.
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP210 Analog VOIP (Site E)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comment**
421		1	

Call Completion Summaries

Site A calling Site C, Method 1

1. Dial 101.
2. Hear dial tone from Site B.
3. Dial 7175662.
4. Await completion. Talk.

Site A calling Site C, Method 2

1. Dial 101#7175662
2. Await completion. Talk.

Note: Some analog VOIP gateways will allow completion by Method 2. Others will not.

Site C calling Site A

1. Dial 7175000.
2. Hear dial tone from Site B VOIP.
3. Dial 201.
4. Await completion. Talk.

Site D calling Site C

1. Dial 9,15077175662.
2. "9" gets outside line. On some PBXs, an "8" may be used to direct calls to the VOIP, while "9" directs calls to the PSTN. However, some PBX units can be programmed to identify the destination patterns of all calls to be directed to the VOIP.
3. PBX at Site D is programmed to divert all calls made to the 507 area code and exchange 717 into the VOIP network. (It would also be possible to divert all calls to all phones in area code 507 into the VOIP network, but it may not be desirable to do so.)
4. The MVP2410 removes the prefix "1507" and adds the prefix "101#" for compatibility with the analog MultiVOIP's phonebook scheme. The "#" is a delimiter separating the analog VOIP's phone number from the digits that the analog VOIP must dial onto its local PSTN to complete the call. The digits "101#7175662" are forwarded to the Site B analog VOIP.
5. The call passes through the IP network (in this case, the Internet).
6. The call arrives at the Site B VOIP. This analog VOIP receives this dialing string from the MVP2410: 101#7175662. The analog VOIP, seeing the "101" prefix, uses its own channel #1 (an FXO port) to connect the call to the PSTN. Then the analog VOIP dials its local phone number 7175662 to complete the call.

Site D calling Site F

A voip call from Pierre PBX to extension 7424 on the key telephone system in Lincoln, Nebraska.

A. The required entry in the Pierre Outbound Phonebook to facilitate origination of the call, would be 1402263742. The call would be directed to the Lincoln voip's IP address, 200.2.9.5.
(Generally on such a call, the caller would have to dial an initial "9." But typically the PBX would not pass the initial "9" to the voip. If the PBX *did* pass along that "9" however, its removal would have to be specified in the local Outbound Phonebook.)

B. The corresponding entry in the Lincoln Inbound Phonebook to facilitate completion of the call would be

1402263742	for calls within the office at Lincoln
1402	for calls to the Lincoln local calling area (PSTN).

Call Event Sequence

1. Caller at Pierre dials 914022637424.
2. Pierre PBX removes "9" and passes 14022637424 to voip.
3. Pierre voip passes remaining string, 14022637424 on to the Lincoln voip
   at IP address 200.2.9.5.
4. The dialed string matches an inbound phonebook entry at the Lincoln voip, namely 1402263742.
5. The Lincoln voip rings one of the three FXS ports connected to the Lincoln
   key phone system.
6. The call will be routed to extension 7424 either by a human receptionist/
   operator or to an auto-attendant (which allows the caller to specify the
   extension to which they wish to be connected).

Site F calling Site D

A voip call from a Lincoln key extension to extension 3117 on the PBX in Pierre, South Dakota.

A. The required entry in the Lincoln Outbound Phonebook to facilitate origination of the call, would be “31”. The string “1615492” would have to be added as a prefix. The call would be directed to the Pierre voip’s IP address, 200.2.9.9.

B. The corresponding entry in the Pierre Inbound Phonebook to facilitate completion of the call would be 1615492.

1. Caller at Lincoln picks up phone receiver, presses button on key phone set. This button has been assigned to a particular voip channel (any one of the three FXS ports).
2. The caller at Lincoln hears dial tone from the Lincoln voip.
3. The caller at Lincoln dials 3117.
4. The Lincoln voip adds the prefix 1615492 and sends the entire dialing string, 16154923117, to the Pierre voip at IP address 200.2.9.9.
5. The Pierre voip matches the called digits 16154923117 to its Inbound Phonebook entry “1615492” .
6. The Pierre PBX dials extension 3117 in the office at Pierre.

Variations in PBX Characteristics

The exact dialing strings needed in the Outbound and Inbound Phonebooks of the MVP2410 will depend on the capabilities of the PBX. Some PBXs require trunk access codes (like an “8” or “9” to access an outside line or to access the VOIP network). Other PBXs can automatically distinguish between intra-PBX calls, PSTN calls, and VOIP calls.

Some PBX units can also insert digits automatically when they receive certain dialing strings from a phone station. For example, a PBX may be programmable to insert automatically the three-digit VOIP identifier strings into calls to be directed to analog VOIPs.

The MVP2410 offers complete flexibility for inter-operation with PBX units so that a coherent dialing scheme can be established to connect a company’s multiple sites together in a way that is convenient and intuitive for phone users. When working together with modern PBX units, the presence of the MVP2410 can be completely transparent to phone users within the company.

Chapter 7: E1 Phonebook Configuration

(European Telephony Standards)

E1 Versus T1 Telephony Environments

We present separate chapters for the MVP3010 MultiVOIP (this chapter) and the MVP2410 MultiVOIP (Chapter 6) because the respective telephony environments in which they operate have different standards and conventions. The MVP3010 is designed to operate under European or E1 standards; the MVP2410 is designed to operate under North American or T1 standards. The configuration of the phonebook is the same in either case. However, differences in the telephony environment give rise to different examples in each case. Series II analog MultiVOIP units (MVP130, MVP130FXS, MVP210, MVP410, and MVP810) can be operated in either the T1 or E1 environments. The examples in this chapter show these analog voip units being used in the same system as the MVP3010 digital MultiVOIP.

E1-Standard Inbound and Outbound MultiVOIP Phonebooks

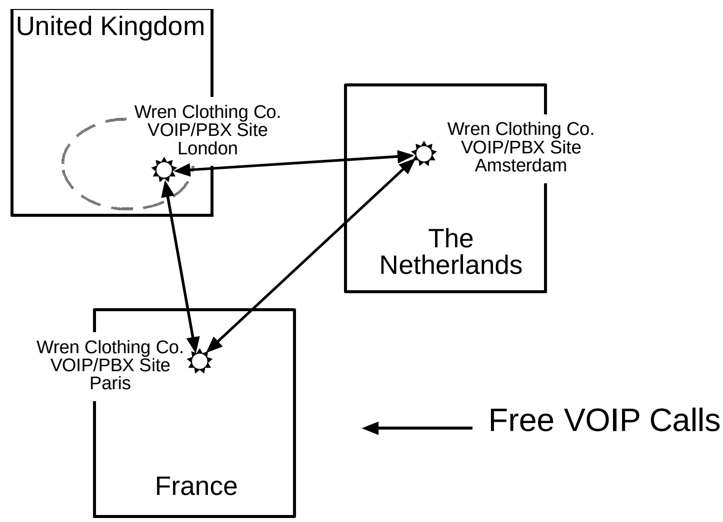
Important Definition:	The MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook describes the dialing sequences that can be used to call that MultiVOIP and how those calls will be directed.

When a VOIP serves a PBX system, the operation of the VOIP should be transparent to the telephone end user and savings in long-distance calling charges should be enjoyed. Use of the VOIP should not require the dialing of extra digits to reach users elsewhere on the VOIP network. On the contrary, VOIP service more commonly reduces dialed digits by allowing users (served by PBXs in facilities in distant cities) to dial their co-workers with 3-, 4-, or 5-digit extensions -- as if they were in the same facility. More importantly, the VOIP system should be configured to maximize savings in long-distance calling charges. To achieve both of these objectives, ease of use and maximized savings, the VOIP phonebooks must be set correctly.

NOTE: VOIPs are commonly used for another reason, as well: VOIPs allow an organization to integrate phone and data traffic onto a single network. Typically these are private networks.

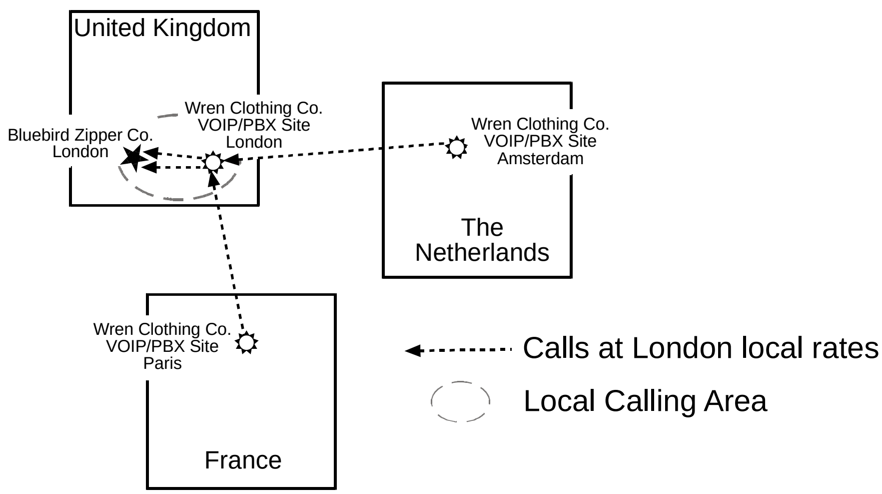
Free Calls: One VOIP Site to Another

The most direct use of the VOIP system is making calls between the offices where the VOIPs are located. Consider, for example, the Wren Clothing Company. This company has VOIP-equipped offices in London, Paris, and Amsterdam, each served by its own PBX. VOIP calls between the three offices completely avoid international long-distance charges. These calls are free. The phonebooks can be set up to allow all Wren Clothing employees to contact each other using 3-, 4-, or 5-digit numbers, as though they were all in the same building.

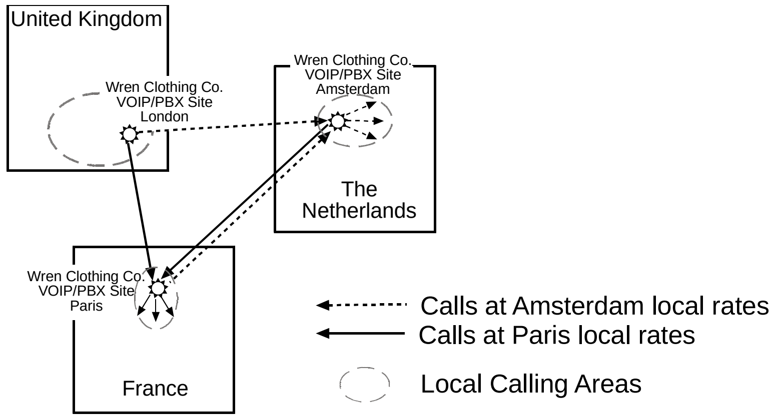


Local Rate Calls: Within Local Calling Area of Remote VOIP

In the second use of the VOIP system, the local calling area of each VOIP location becomes accessible to all of the VOIP system's users. As a result, international calls can be made at local calling rates. For example, suppose that Wren Clothing buys its zippers from The Bluebird Zipper Company in the western part of metropolitan London. In that case, Wren Clothing personnel in both Paris and Amsterdam could call the Bluebird Zipper Company without paying international long-distance rates. Only London local phone rates would be charged. This applies to calls completed anywhere in London's local calling area (which includes both Inner London and Outer London). Generally, local calling rates apply only within a single area code, and, for all calls outside that area code, national rates apply. There are, however, some European cases where local calling rates extend beyond a single area code. Local rates between Inner and Outer London are one example of this. (It is also possible, in some locations, that calls within an area code may be national calls. But this is rare.)

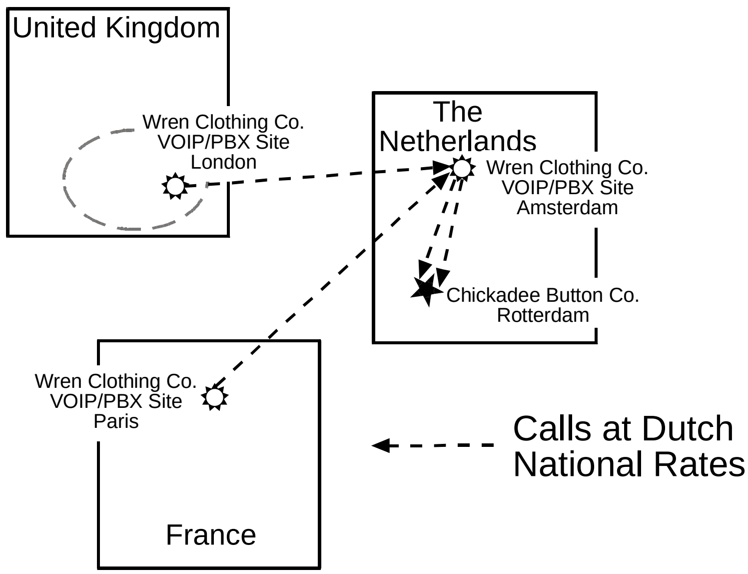


Similarly, the VOIP system allows Wren Clothing employees in London and Amsterdam to call anywhere in Paris at local rates; it allows Wren Clothing employees in Paris and London to call anywhere in Amsterdam at local rates.

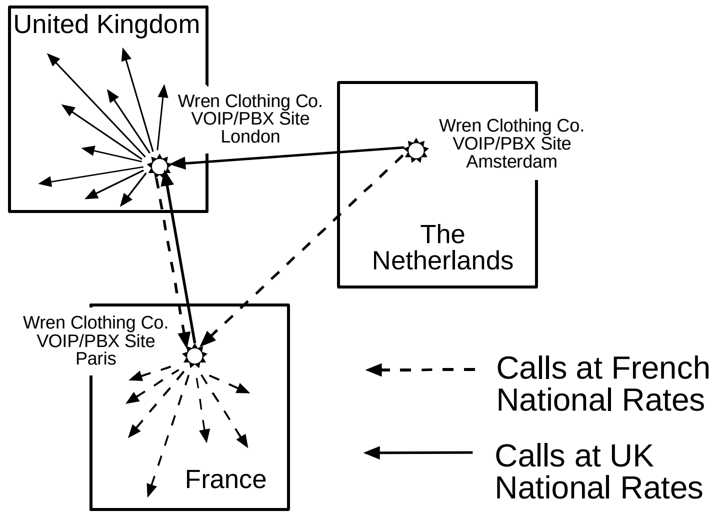


National Rate Calls: Within Nation of Remote VOIP Site

In the third use of the VOIP system, the national calling area of each VOIP location becomes accessible to all of the VOIP system's users. As a result, international calls can be made at national calling rates. Again, significant savings are possible. For example, suppose that the Wren Clothing Company buys its buttons from the Chickadee Button Company in the Dutch city of Rotterdam. In that case, Wren Clothing personnel in both London and Paris could call the Chickadee Button Company without paying international long-distance rates; only Dutch national calling rates would be charged. This applies to calls completed anywhere in The Netherlands.



Similarly, the VOIP system allows Wren Clothing employees in London and Amsterdam to call anywhere in France at French national rates; it allows Wren Clothing employees in Paris and Amsterdam to call anywhere in the United Kingdom at its national rates.



Inbound versus Outbound Phonebooks

To make the VOIP system transparent to phone users and to allow all possible free and reduced-rate calls, the VOIP administrator must configure the "Outbound" and "Inbound" phone-books of each VoIP in the system.

The "Outbound" phonebook for a particular VOIP unit describes the dialing sequences required for a call to originate locally (typically in a PBX in a particular facility) and reach any of its possible destinations at remote VOIP sites, including calls terminating at points beyond the remote VOIP site.

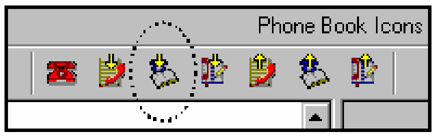
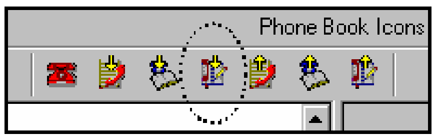
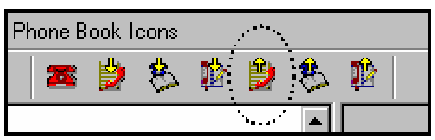
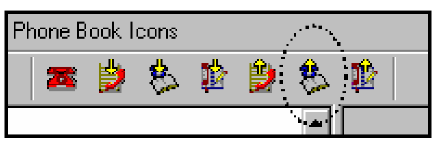
The "Inbound" phonebook for a particular VOIP unit describes the dialing sequences required for a call to originate remotely from any other VOIP sites in the system, and to terminate on that particular VOIP.

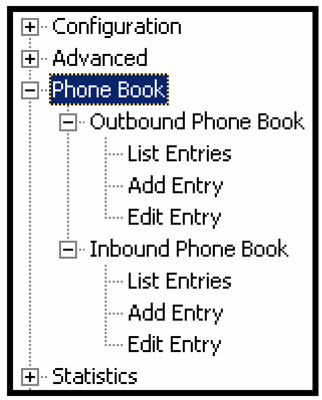
Briefly stated, *the MultiVOIP's Outbound phonebook lists the phone stations it can call; its Inbound phonebook lists the dialing sequences that can be used to call that MultiVOIP*. (Of course, the phone numbers are not literally "listed" individually.) The phone stations that can originate or complete calls over the VOIP system are described by numerical rules called "destination patterns." These destination patterns generally consist of country codes, area codes or city codes, and local phone exchange numbers.

In order for any VOIP phone call to be made, there must be both an Inbound Phonebook entry and an Outbound Phonebook entry that describe the end-to-end connection. The phone station originating the call must be connected to the VOIP system. The Outbound Phonebook for that VOIP unit must have a destination pattern entry that includes the 'called' phone (that is, the phone completing the call). The Inbound Phonebook of the VOIP where the call is completed must have a destination pattern entry that includes the digit sequence dialed by the originating phone station.

The PhoneBook Configuration procedure below is brief, but it is followed by an example case. For many people, the example case may be easier to grasp than the procedure steps. Configuration is not difficult, but all phone number sequences, destination patterns, and other information must be entered exactly; otherwise connections will not be made.

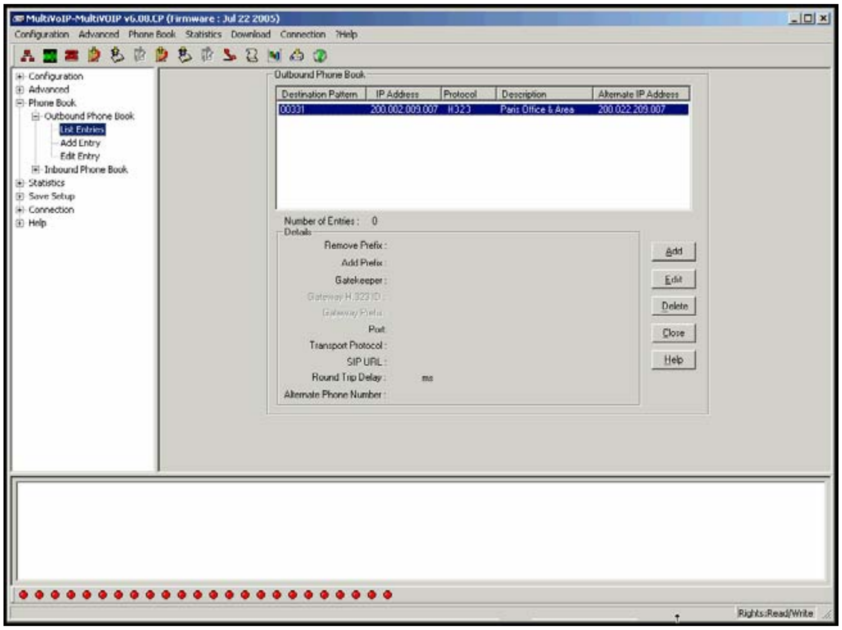
Phonebook configuration screens can be accessed using icons or the sidebar menu.

Phonebook Icons	Description
	Phonebook Configuration
	Inbound Phonebook Entries List
	Add Inbound Phonebook Entry
	Edit selected Inbound Phonebook Entry
	Outbound Phonebook Entries List
	Add Outbound Phonebook Entry
	Edit selected Outbound Phonebook Entry

<table>
<tr><td>Phonebook Pulldown Menu</td><td></td></tr>
<tr><td colspan="2">
</td></tr>
<tr><td>Inbound Phonebook Shortcut</td><td>Outbound Phonebook Shortcut</td></tr>
<tr><td>Alt + I</td><td>Alt + O</td></tr>
<tr><td>Phonebook Sidebar Menu</td><td></td></tr>
<tr><td colspan="2">
</td></tr>
</table>

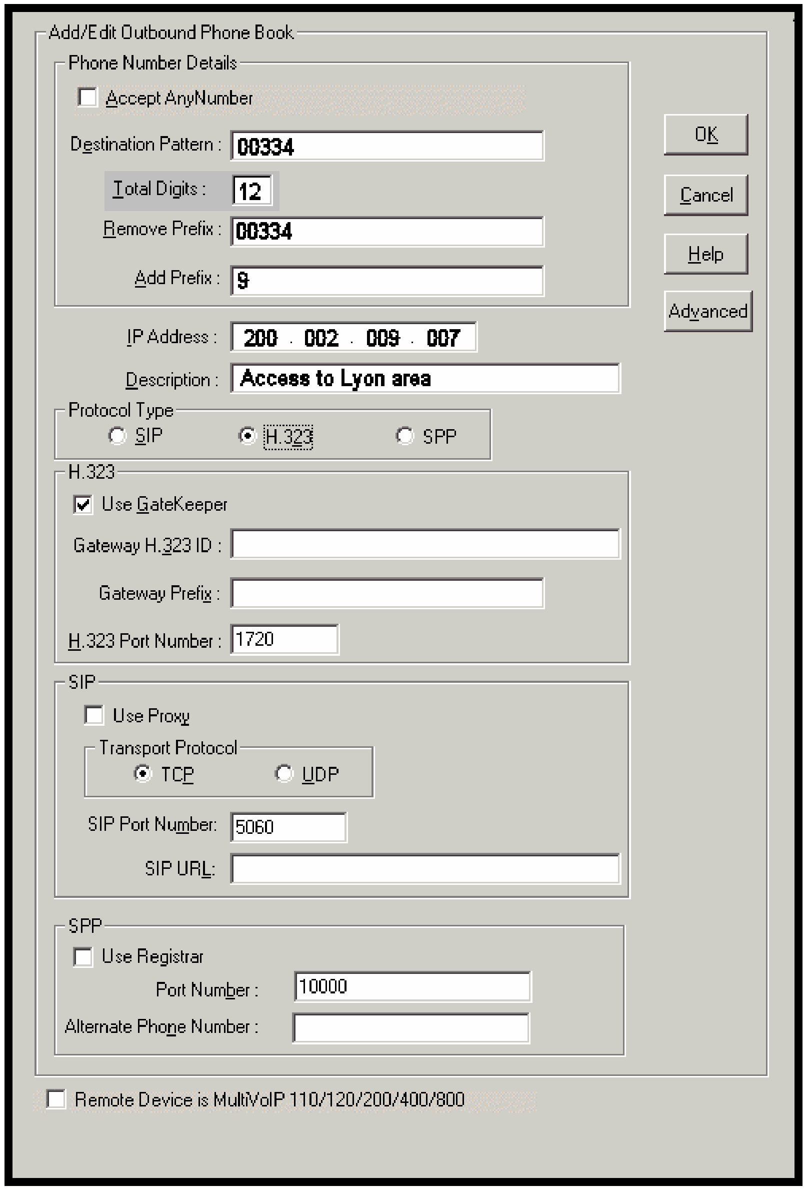
Phonebook Configuration Procedure

1. Select **Outbound Phone Book/List Entries**.



Click **Add**.

2. The **Add/Edit Outbound PhoneBook** screen appears.



Enter Outbound PhoneBook data for your MultiVOIP unit. Note that the Advanced button gives access to the Alternate IP Routing feature, if needed. Alternate IP Routing can be implemented in a secondary screen (as described after the primary screen field definitions below).

The fields of the **Add/Edit Outbound Phone Book** screen are described in the table below.

Add/Edit Outbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Accept Any Number	Y/N	When checked, "Any Number" appears as the value in the Destination Pattern field. The **Any Number** feature works differently depending on whether or not an external routing device is used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Any Number** is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the **IP Address** in the Add/Edit Outbound Phone Book screen. "Any Number" can be used in addition to one or more Destination Patterns. **When external routing device is used**. If Any Number is selected, calls to phone numbers not matching a listed Destination Pattern will be directed to the external routing device used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). The IP Address of the external routing device must be set in the Phone Book Configuration screen.

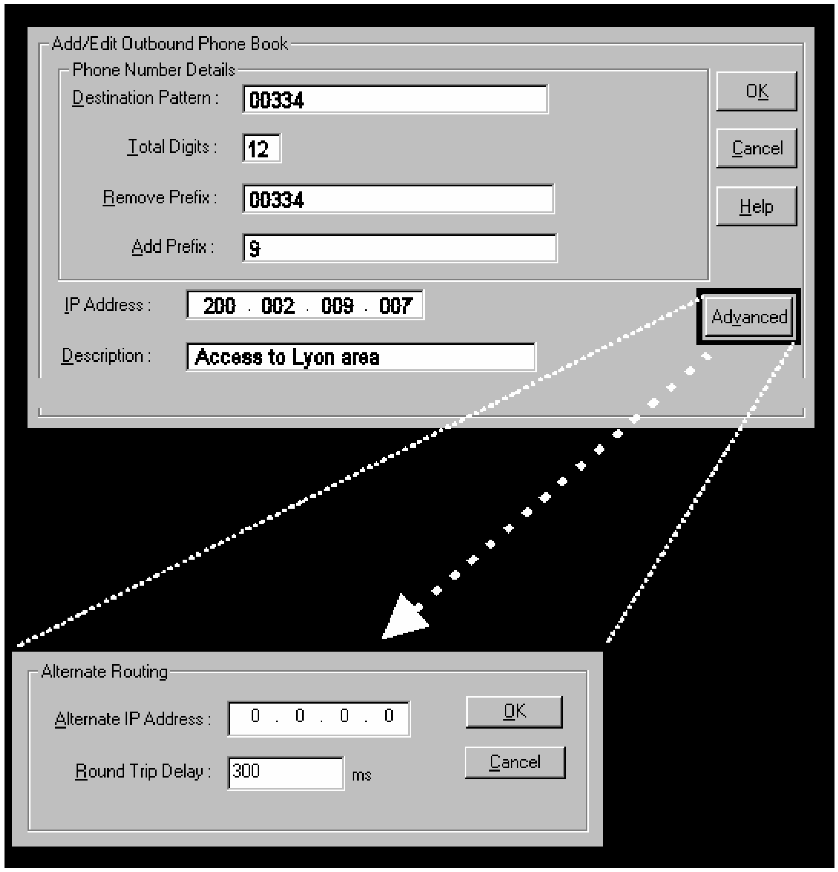
Add/Edit Outbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Destination Pattern	prefixes, area codes, exchanges, line numbers, extensions	Defines the beginning of dialing sequences for calls that will be connected to another VOIP in the system. Numbers beginning with these sequences are diverted from the PTSN and carried on Internet or other IP network.
Total Digits	as needed	number of digits the phone user must dial to reach specified destination
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination
Add Prefix	dialed digits	digits to be added before completing call to destination
IP Address	n.n.n.n for = 0-255	the IP address to which the call will be directed if it begins with the destination pattern given
Description	alpha-numeric	Describes the facility or geographical location at which the call will be completed.
Protocol Type	SIP, H.323, or SPP	Indicates protocol to be used in outbound transmission.

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
H.323 fields		
Use Gatekeepr	Y/N	Indicates whether or not gatekeeper is used.
Gateway H.323 ID	alpha-numeric	The H.323 ID assigned to the destination MultiVOIP. Only valid if "Use Gatekeeper" is enabled for this entry.
Gateway Prefix	numeric	This number becomes registered with the GateKeeper. Call requests sent to the gatekeeper and preceded by this prefix will be routed to the VOIP gateway.
H.323 Port Number	1720	This parameter pertains to Q.931, which is the H.323 call signaling protocol for setup and termination of calls (aka ITU-T Recommendation I.451). H.323 employs only one "well-known" port (1720) for Q.931 signaling. If Q.931 message-oriented signaling protocol is used, the port number 1720 must be chosen.

Add/Edit Outbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
SIP Fields		
Use Proxy	Y/N	Select if proxy server is used.
Transport Protocol	TCP or UDP	Voip administrator must choose between UDP and TCP transmission protocols. UDP is a high-speed, low-overhead connectionless protocol where data is transmitted without acknowledgment, guaranteed delivery, or guaranteed packet sequence integrity. TCP is slower connection-oriented protocol with greater overhead, but having acknowledgment and guarantees delivery and packet sequence integrity.
SIP Port Number	5060 or other *See RFC3087 ("Control of Service Context using SIP Request-URI," by the Network Working Group).	The SIP Port Number is a UDP logical port number. The voip will "listen" for SIP messages at this logical port. If SIP is used, 5060 is the default, standard, or "well known" port number to be used. If 5060 is not used, then the port number used is that specified in the SIP Request URI (Universal Resource Identifier).
SIP URL	*sip.userphone* @ *hostserver*, where "userphone" is the telephone number and "hostserver" is the domain name or an address on the network	Looking similar to an email address, a **SIP URL** identifies a user's address. In SIP communications, each caller or callee is identified by a SIP url: sip:user_name@host_name. The format of a sip url is very similar to an email address, except that the "sip:" prefix is used.

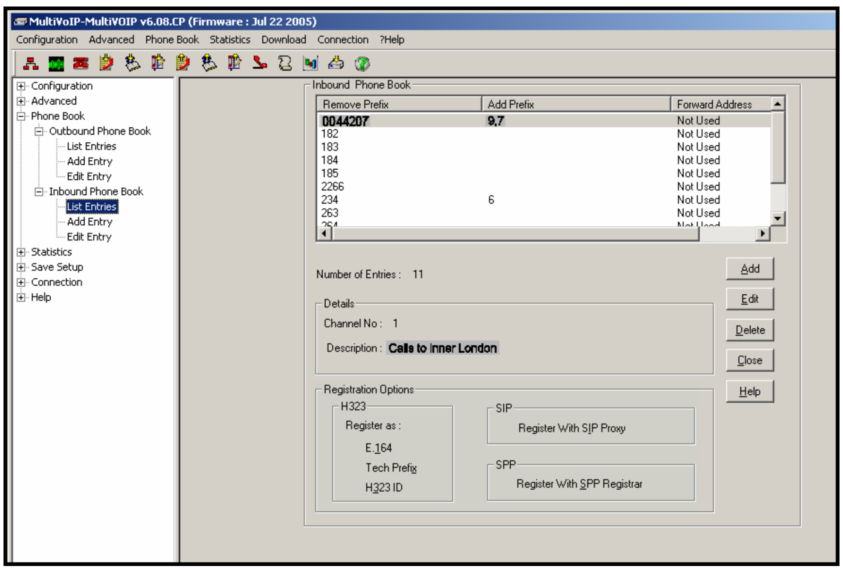
Add/Edit Outbound Phone Book: Field Def'ns (cont'd)		
Field Name	**Values**	**Description**
SPP Fields		
Use Registrar	**Values:** Y/N **Description:** Select this checkbox to use registrar when voip system is operating in the "Registrar/Client" SPP mode. In this mode, one voip (the registrar, as set in Phonebook Configuration screen) has a static IP address and all other voips (clients) point to the registar's IP address as functionally their own. However, if your voip system overall is operating in "Registrar/Client" mode but you want to make an exception and use Direct mode for the destination pattern of this particular Add/Edit Phonebook entry, leave this checkbox unselected. Leave this checkbox unselected if your overall voip system is operating in the "Direct" SPP mode. In this mode, all voips in system are peers and each has its own static IP address.	
Port Number	**Values:** numeric **Description:** When operating in "Registrar/Client" mode, this is the port by which the gateway receives all SPP data and control messages from the registrar gateway. (This ability to receive all data and messages via one port allows the voip to operate behind a firewall with only one port open.) When operating in "Direct" mode, this is the Port by which peer voips receive data and messages.	
Alternate Phone Number	numeric	Phone number associated with alternate IP routing.
Remote Device is …	Y/N	Check when system includes 1st-generation MultiVOIPs to allow inter-operation. These include MVP-110/120/200/400/800 MultiVOIP units.
Advanced button	**Values:** N/A **Description:** Gives access to secondary screen where an Alternate IP Route can be specified for backup or redundancy of signal paths. See discussion on next page. For SIP & H.323 operation only.	

Clicking on the **Advanced** button brings up the **Alternate Routing** secondary screen. This feature provides an alternate path for calls if the primary IP network cannot carry the traffic. Often in cases of failure, call traffic is temporarily diverted into the PSTN. However, this feature could also be used to divert traffic to a redundant (backup) unit in case one voip unit fails. The user must specify the IP address of the alternate route for each destination pattern entry in the Outbound Phonebook.

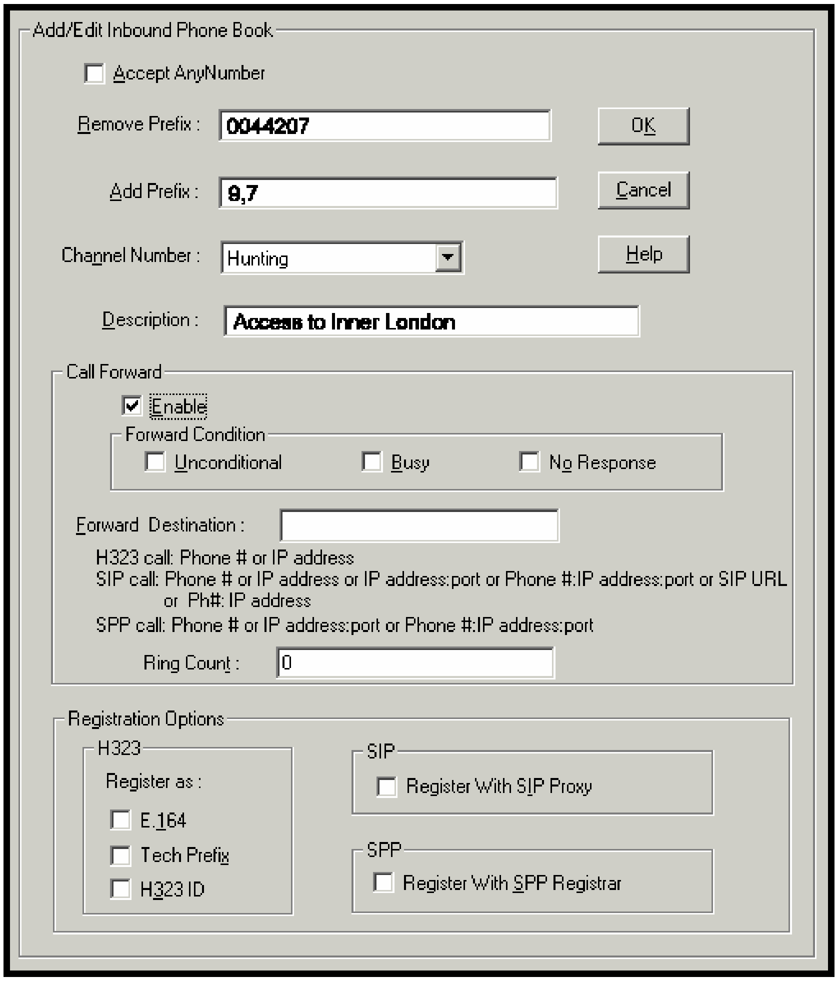


Alternate Routing Field Definitions		
Field Name	**Values**	**Description**
Alternate IP Address	n.n.n.n where n= 0-255	Alternate destination for outbound data traffic in case of excessive delay in data transmission.
Round Trip Delay	milliseconds	The Round Trip Delay is the criterion for judging when a data pathway is considered blocked. When the delay exceeds the threshold specified here, the data stream will be diverted to the alternate destination specified as the Alternate IP Address.

3. Select **Inbound PhoneBook/List Entries**.



4. The **Add/Edit Inbound PhoneBook** screen appears.



Enter Inbound PhoneBook data for your MultiVOIP unit. The fields of the Add/Edit Inbound PhoneBook screen are described in the table below.

Add/Edit Inbound Phone Book: Field Definitions		
Field Name	**Values**	**Description**
Accept Any Number	Y/N	When checked, "Any Number" appears as the value in the Remove Prefix field. The **Any Number** feature of the Inbound Phone Book does not work when an external routing device is used (Gatekeeper for H323 protocol, Proxy for SIP protocol, Registrar for SPP protocol). **When no external routing device is used**. If **Any Number** is selected, calls received from phone numbers not matching a listed Prefix (shown in the Remove Prefix column of the Inbound Phone Book) will be admitted into the voip on the channel listed in the **Channel Number** field. "Any Number" can be used in addition to one or more Prefixes.
Remove Prefix	dialed digits	portion of dialed number to be removed before completing call to destination (often a local PBX)
Add Prefix	dialed digits	digits to be added before completing call to destination (often a local PBX)

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Channel Number	1-30, or "Hunting"	E1 channel number to which the call will be assigned as it enters the local telephony equipment (often a local PBX). "Hunting" directs the call to any available channel.
Description	--	Describes the facility or geographical location at which the call originated.
Call Forward Parameters		
Enable	Y/N	Click the check-box to enable the call-forwarding feature.
Forward Condition	Uncondit.; Busy No Resp.	**Unconditional**. When selected, all calls received will be forwarded. **Busy**. When selected, calls will be forwarded when station is busy. **No Response**. When selected, calls will be forwarded if called party does not answer after a specified number of rings, as specified in **Ring Count** field. Forwarding can be conditioned on both "Busy" and "No Response."

Add/Edit Inbound Phone Book: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Forward Destination IP address, phone number, port number, etc.	Phone number or IP address to which calls will be directed. For H.323 calls, the Forward Destination can be either a Phone Number of an IP Address. For SIP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address, (c) IP address: port number, (d) phone number:IP addr: port number, (e) SIP URL, or (f) phone #: IP address. For SPP calls, the Forward Destination can be one of the following: (a) phone number, (b) IP address: port, or (c) phone number: IP address: port.	
Ring Count	integer	When No Response is condition for forwarding calls, this determines how many unanswered rings are needed to trigger the forwarding.
Registration Option Parameters	In an H.323 voip system, gateways can register with the system using one of these identifiers: (a) an E.164 identifier, (b) a Tech Prefix identifier, or (c) an H.323 ID identifier. In a SIP voip system, gateways can register with the SIP Proxy. In an SPP voip system, gateways can register with the SPP Registrar voip unit.	

5. When your Outbound and Inbound PhoneBook entries are completed, click on **Save Setup** in the sidebar menu to save your configuration.

   You can change your configuration at any time as needed for your system.

Remember that the initial MultiVOIP setup must be done locally or via the built-in Remote Configuration/Command Modem using the MultiVOIP program. However, after the initial configuration is complete, all of the MultiVOIP units in the VOIP system can be configured, re-configured, and updated from one location using the MultiVOIP web GUI software program or the MultiVOIP program (in conjunction with the built-in modem).

E1 Phonebook Examples

To demonstrate how Outbound and Inbound PhoneBook entries work in an international VOIP system, we will re-visit our previous example in greater detail. It's an international company with offices in London, Paris, and Amsterdam. In each office, a MVP3010 has been connected to the PBX system.

3 Sites, All-E1 Example

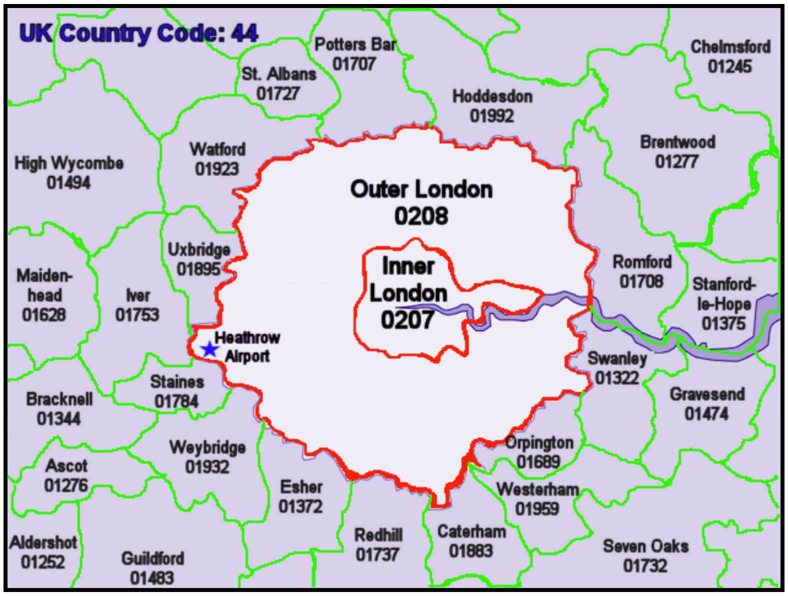
The VOIP system will have the following features:

1. Employees in all cities will be able to call each other over the VOIP system using 4-digit extensions.

2. Calls to Outer London and Inner London, greater Amsterdam, and greater Paris will be accessible to all company offices as local calls.

3. Vendors in Guildford, Lyon, and Rotterdam can be contacted as national calls by all company offices.

Note that the phonebook entries for Series II analog MultiVOIPs (MVP-210/410/810) used in Euro-type telephony settings will be the same in format as entries for the MVP3010.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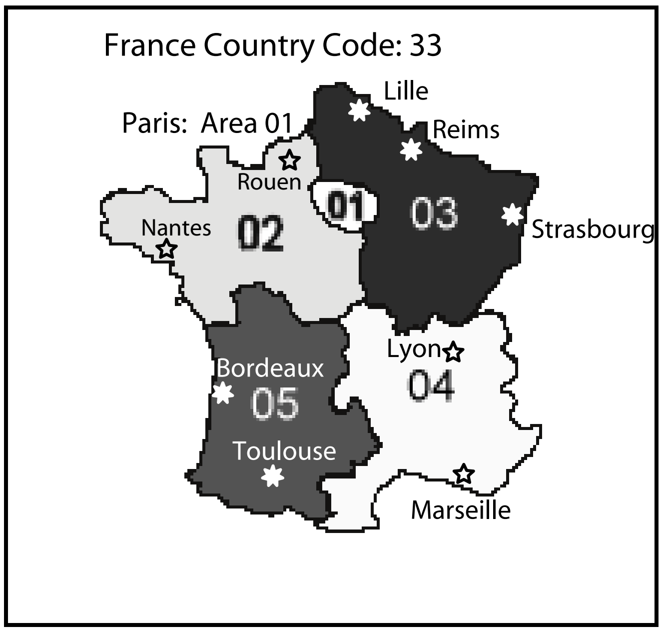
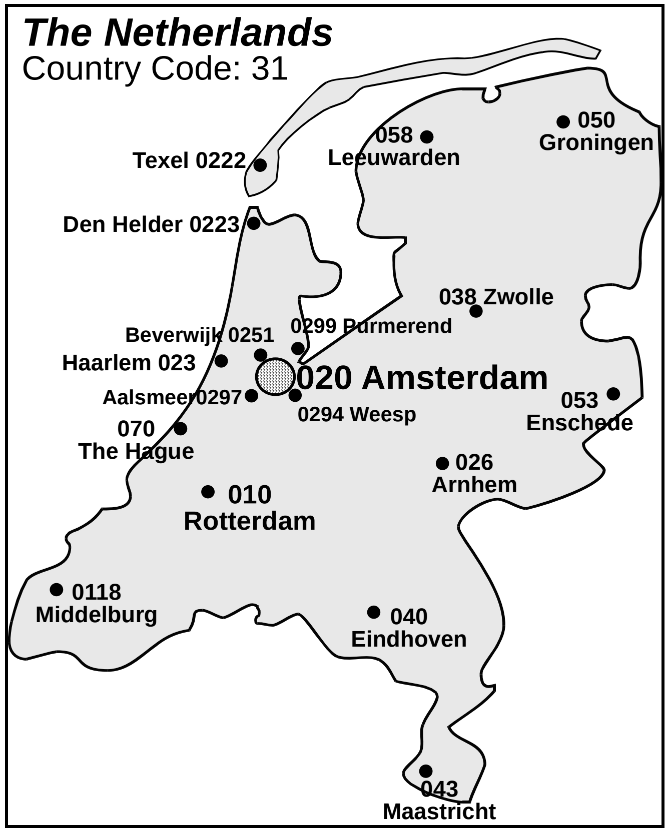
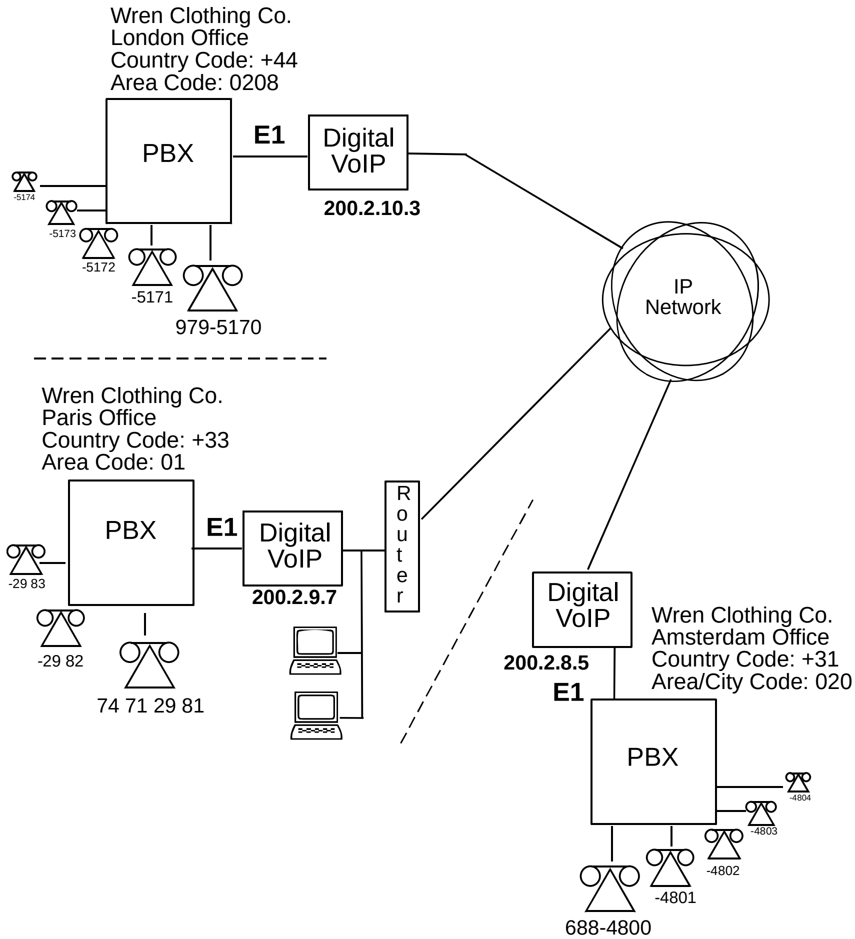
Potters Bar 01707
St. Albans 01727
Chelmsford 01245
Hoddesdon 01992
Brentwood 01277
Watford 01923
High Wycombe 01494
Outer London 0208
Uxbridge 01895
Inner London 0207
Romford 01708
Maiden-head 01628
Iver 01753
Stanford-le-Hope 01375
Heathrow Airport
Swanley 01322
Staines 01784
Bracknell 01344
Gravesend 01474
Orpington 01689
Weybridge 01932
Ascot 01276
Esher 01372
Westerham 01959
Caterham 01883
Redhill 01737
Aldershot 01252
Seven Oaks 01732
Guildford 01483

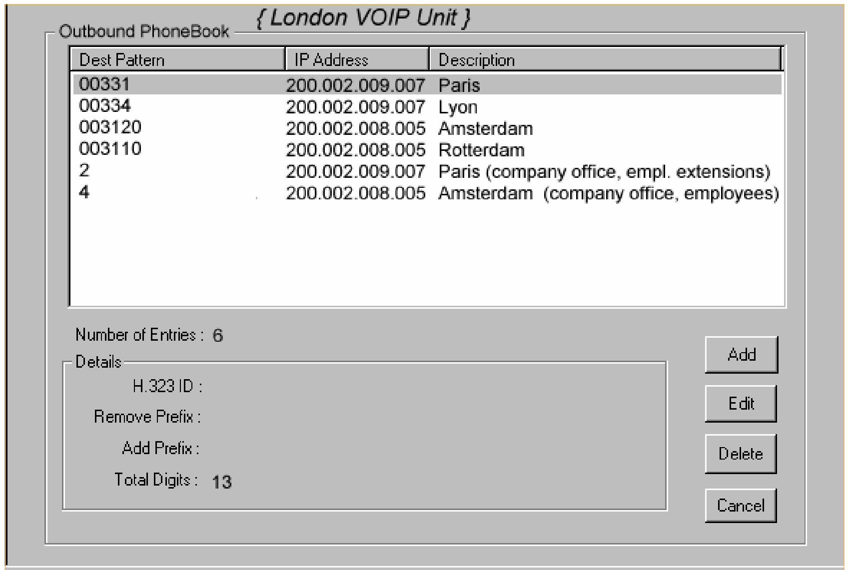

France Country Code: 33
Lille
Paris: Area 01
Reims
Rouen
01
03
Strasbourg
Nantes
02
Lyon
04
Bordeaux
05
Toulouse
Marseille


The Netherlands
Country Code: 31
Texel 0222
058 Leeuwarden
050 Groningen
Den Helder 0223
038 Zwolle
Beverwijk 0251
0299 Purmerend
Haarlem 023
020 Amsterdam
Aalsmeer 0297
0294 Weesp
053 Enschede
070 The Hague
026 Arnhem
010 Rotterdam
0118 Middelburg
040 Eindhoven
043 Maastricht

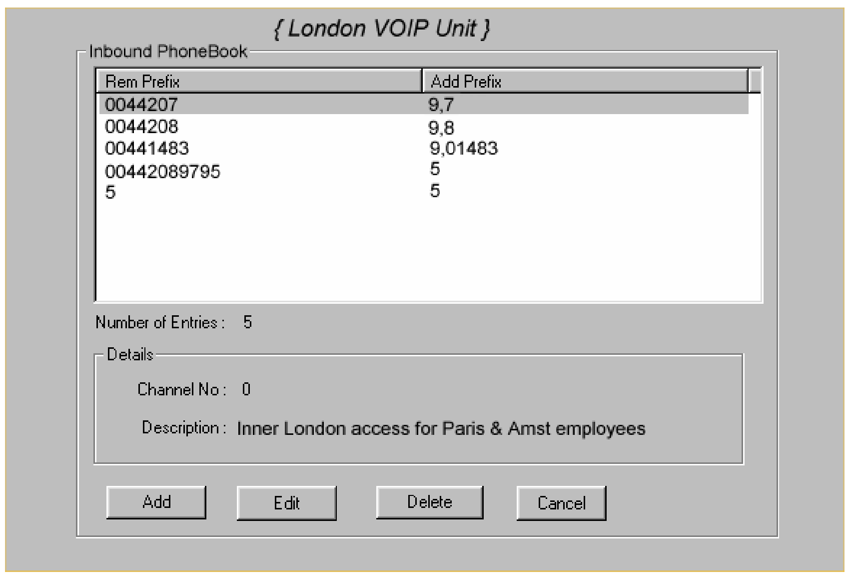
An outline of the equipment setup in these three offices is shown below.



The screen below shows Outbound PhoneBook entries for the VOIP located in the company's London facility

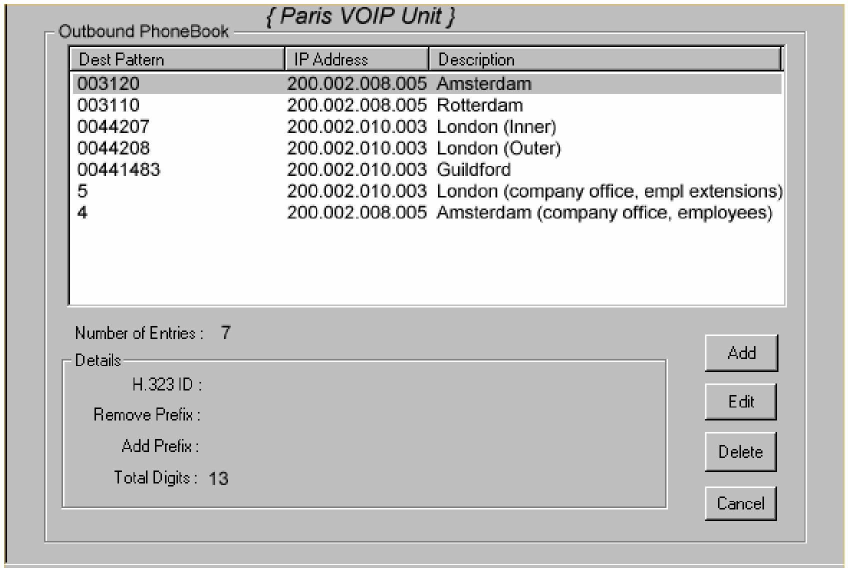


The Inbound PhoneBook for the London VOIP is shown below.

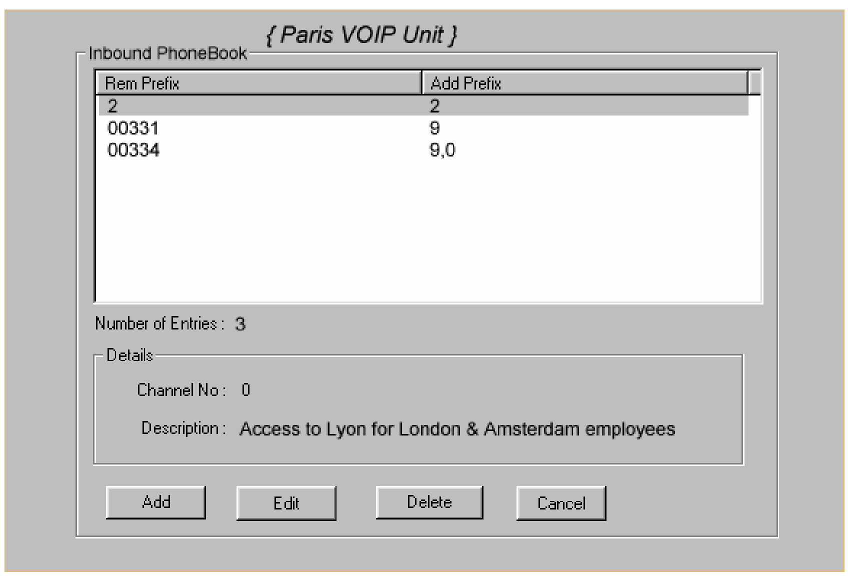


NOTE: Commas are allowed in the Inbound Phonebook, but **not** in the Outbound Phonebook. Commas denote a brief pause for a dial tone, allowing time for the PBX to get an outside line.

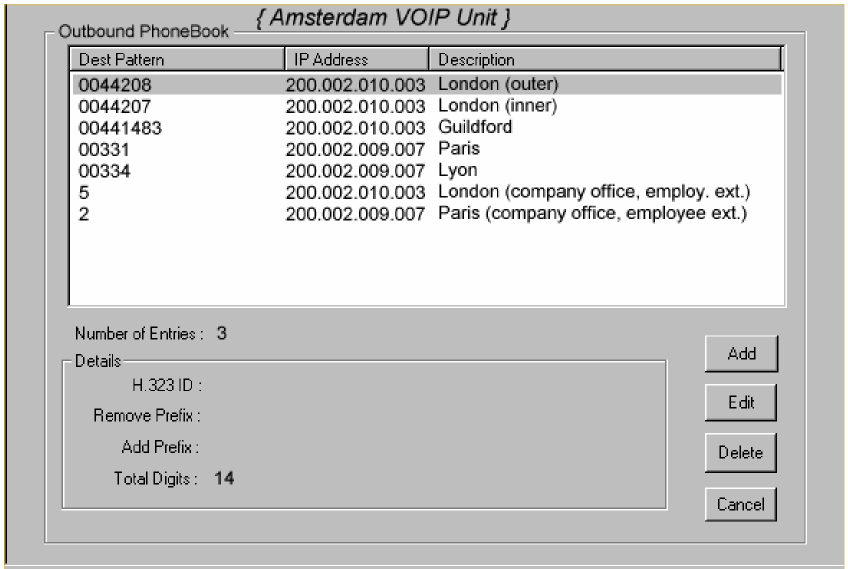
The screen below shows Outbound PhoneBook entries for the VOIP located in the company's Paris facility.



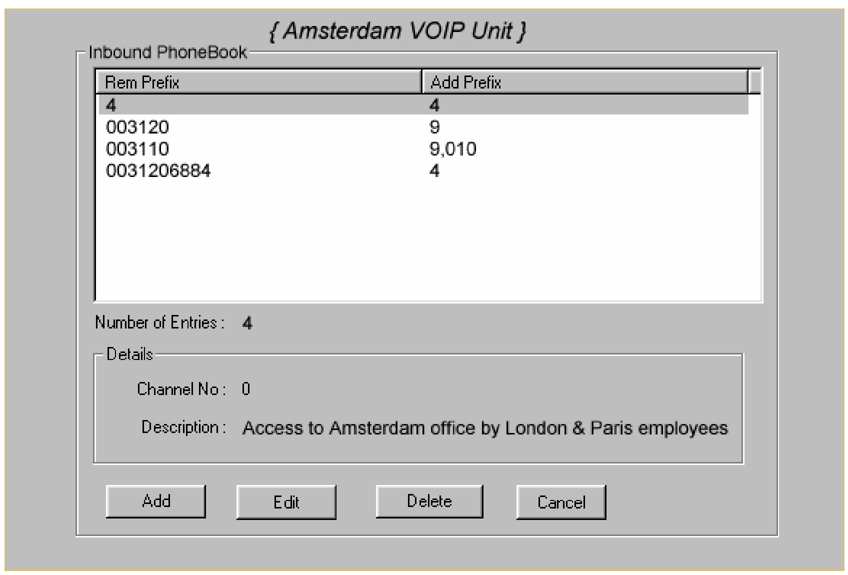
The Inbound PhoneBook for the Paris VOIP is shown below.



The screen below shows Outbound PhoneBook entries for the VOIP in the company's Amsterdam facility.

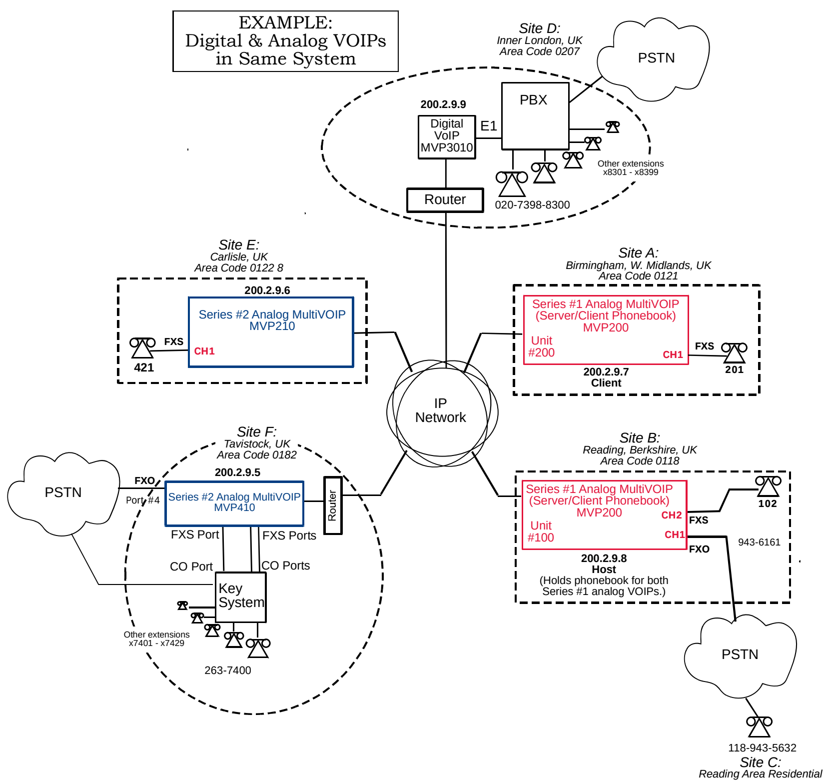


The Inbound PhoneBook for the Amsterdam VOIP is shown below.



Configuring Digital & Analog VOIPs in Same System

Analog MultiVOIP units, like the MVP-210/410/810 are compatible with digital MultiVOIP units like the MVP3010. In many cases, digital and analog VOIP units will appear in the same telephony/IP system. In addition to MVP-210/410/810 MultiVOIP units (Series II units), legacy analog VOIP units (Series I units made by MultiTech) may be included in the system, as well. When legacy VOIP units are included, the VOIP administrator must handle two styles of phonebooks in the same VOIP network. The diagram below shows a small-scale system of this kind: one digital VOIP (the MVP3010) operates with two Series II analog VOIPs (an MVP210 and an MVP410), and two Series I legacy VOIPs (two MVP200 units).



The Series I analog VOIP phone book resides in the "Host" VOIP unit at Site B. It applies to both of the Series I analog VOIP units.

Each of the Series II analog MultiVOIPs (the MVP210 and the MVP410) requires its own inbound and outbound phonebooks. The MVP3010 digital MultiVOIP requires its own inbound and outbound phonebooks, as well.

These **seven** phone books are shown below.

Phone Book for Analog VOIP Host Unit (Site B)			
VOIP Dir # -OR- Destination Pattern	IP Address	Channel	Comments
102	200.2.9.8	2	Site B, FXS channel. (Reading, UK)
101	200.2.9.8	1	Site B, FXO channel. (Reading, UK)
201	200.2.9.7	1	Site A, FXS channel. (Birmingham)
421	200.2.9.6	0	Site E, FXS channel. (Carlisle, UK)
018226374 Note 3.	200.2.9.5	0	Gives remote voip users access to key phone system extensions at Tavistock office (Site F). The key system might be arranged either so that calls go through a human operator or through an auto-attendant (which prompts user to dial the desired extension).
0182	200.2.9.5	4	Gives remote voip users access to Tavistock PSTN via FXO port (#4) at Site F.
3xx	200.2.9.9	0 (Note 1.)	Allows remote voip users to call all PBX extensions at Site D (Inner London) using only three digits.

Phone Book for Analog VOIP Host Unit (Site B) (continued)			
VOIP Dir # -OR- Destination Pattern	IP Address	Channel	Comments
0207 xxx xxxx	200.2.9.9	0 (Note 2.)	Gives remote voip users access to phone numbers in 0207 area code (Inner London) in which Site D is located.
0208 xxx xxxx	200.2.9.9	0 (Note 2.)	Gives remote voip users access to phone numbers in 0208 area code (Outer London) for which calls are local from Site D (Inner London).

Note 1. The "x" is a wildcard character.

Note 2. By specifying "Channel 0," we instruct the MVP3010 to choose any available data channel to carry the call.

Note 3. Note that Site F key system has only 30 extensions (x7400-7429). This destination pattern (018226374) actually directs calls to 402-263-74**30** through
402-263-74**99** into the key system, as well.
This means that such calls, which belong on the PSTN, cannot be completed. In some cases, this might be inconsequential because an entire exchange (fully used or not) might have been reserved for the company or it might be unnecessary to reach those numbers. However, to specify only the 30 lines actually used by the key system, the destination pattern 018226374 would have to be replaced by three other destination patterns, namely 018226374**0**, 018226374**1**, and 018226374**2**. In this way, calls to 0182-263-74**30** through 0182-263-74**99** would be properly directed to the PSTN. In the Site D outbound phonebook, the 30 lines are defined exactly, that is, without making any adjacent phone numbers unreachable through the voip system.

The Outbound PhoneBook of the MVP3010 is shown below.

Outbound Phone Book for MVP3010 Digital VOIP (Site D)				
Destin. Pattern	**Remove Prefix**	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Birmingham).
901189	901189	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Reading area using the FXO channel (channel #1) of the Site B VOIP (Reading, UK).
421	--	--	200.2.9.6	Calls to Site E (Carlisle).
90182				Calls to Tavistock local PSTN (Site F) could be arranged by operator or possibly by auto-attendant.
90182 263 740	9	--	200.2.9.5	Calls to extensions of key phone system at Tavistock office.
90182 263 741	9	--	200.2.9.5	
90182 263 742	9	--	200.2.9.5	
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Reading).
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

The Inbound PhoneBook of the MVP3010 is shown below.

Inbound Phone Book for MVP3010 Digital VOIP (Site D)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comments**
0207	9,7 Note 4. Note 5.	0	Allows phone users at remote voip sites to call local numbers (those within the Site D area code, 0207, Inner London) over the VOIP network.
0208	9,8 Note 4. Note 5.	0	Allows phone users at remote voip sites to call local numbers (those in Outer London) over the VOIP network.
0207 39883	3	0	Allows phone users at remote voip sites to call extensions of the Site D PBX using three digits, beginning with “3” .
Note 4. “9” gives PBX station users access to outside line. Note 5. The comma represents a one-second pause, the time required for the user to receive a dial tone on the outside line (PSTN). Commas can be used in the Inbound Phonebook, but **not** in the Outbound Phonebook.			

Outbound Phone Book for MVP410 Analog VOIP (Site F)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Birmingham).
01189	0118	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Reading area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Reading).
421			200.2.9.6	Calls to Site E (Carlisle).
0207			200.2.9.9	Calls to Inner London area PSTN via Site D PBX.
0208			200.2.9.9	Calls to Inner London area PSTN via Site D PBX.
3	--	0207 398 8	200.2.9.9	Calls to Inner London PBX extensions with three digits.
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP410 Analog VOIP (Site F)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comment**
01822	2	4	Calls to Tavistock local PSTN through FXO port (Port #4) at Site F.
0182 263 740	740.	0	Gives remote voip users, access to extensions of key phone system atTavistock office. Because call is completed at key system, abbreviated dialing (3-digits) is not workable. Human operator or auto-attendant is needed to complete these calls.
0182 263 741	741.	0	
0182 263 742	742	0	

Outbound Phone Book for MVP210 Analog VOIP (Site E)				
Destin. Pattern	Remove Prefix	**Add Prefix**	**IP Address**	**Comment**
201			200.2.9.7	To originate calls to Site A (Birmingham).
01189	0118	101# Note 3.	200.2.9.8	To originate calls to any PSTN phone in Reading area using the FXO channel (channel #1) of the Site B VOIP.
102			200.2.9.8	To originate calls to phone connected to FXS port (channel #2) of the Site B VOIP (Reading).
01822	01822	--	200.2.9.5	Calls to Tavistock area PSTN (via FXO channel of the Site F VOIP).
0182 26374			200.2.9.5	Calls to Tavistock key system operator or auto-attendant.
0207	0207		200.2.9.9	Calls to London area PSTN via Site D PBX.
8		0207 398	200.2.9.9	Calls to London PBX extensions with four digits.
Note 3. The pound sign ("#") is a delimiter separating the VOIP number from the standard telephony phone number.				

Inbound Phonebook for MVP210 Analog VOIP (Site E)			
Remove Prefix	**Add Prefix**	**Channel Number**	**Comment**
421		1	

Call Completion Summaries

Site A calling Site C, Method 1

1. Dial 101.
2. Hear dial tone from Site B.
3. Dial 9435632.
4. Await completion. Talk.

Site A calling Site C, Method 2

5. Dial 101#9435632
6. Await completion. Talk.

Note: Some analog VOIP gateways will allow completion by Method 2. Others will not.

Site C calling Site A

1. Dial 9436161.
2. Hear dial tone from Site B VOIP.
3. Dial 201.
4. Await completion. Talk.

Site D calling Site C

1. Dial 901189435632.
2. "9" gets outside line. On some PBXs, an "8" may be used to direct calls to the VOIP, while "9" directs calls to the PSTN. However, some PBX units can be programmed to identify the destination patterns of all calls to be directed to the VOIP.
3. PBX at Site D is programmed to divert all calls made to the 118 area code and exchange 943 into the VOIP network. (It would also be possible to divert *all* calls to all phones in area code 118 into the VOIP network, but it may not be desirable to do so.)
4. The MVP3010 removes the prefix "0118" and adds the prefix "101#" for compatibility with the analog MultiVOIP's phonebook scheme. The "#" is a delimiter separating the analog VOIP's phone number from the digits that the analog VOIP must dial onto its local PSTN to complete the call. The digits "101#9435632" are forwarded to the Site B analog VOIP.
5. The call passes through the IP network (in this case, the Internet).
6. The call arrives at the Site B VOIP. This analog VOIP receives this dialing string from the MVP3010: 101#9435632. The analog VOIP, seeing the "101" prefix, uses its own channel #1 (an FXO port) to connect the call to the PSTN. Then the analog VOIP dials its local phone number 9435632 to complete the call.

NOTE: In the case of Reading, Berkshire,, England, both "1189" and "1183" are considered local area codes. This is, in a sense however, a matter of terminology. It simply means that numbers of the form 9xx-xxxx and 3xx-xxxx are both local calls for users at other sites in the VOIP network.

Site D calling Site F

A voip call from Inner London PBX to extension 7424 on the key telephone system in Tavistock, UK.

A. The required entry in the London Outbound Phonebook to facilitate origination of the call, would be 90182263742. The call would be directed to the Tavistock voip's IP address, 200.2.9.5. (Generally on such a call, the caller would have to dial an initial "9". But typically the PBX would not pass the initial "9" dialed to the voip. If the PBX *did* pass along that "9" however, its removal would have to be specified in the local Outbound Phonebook.)

B. The corresponding entry in the Tavistock Inbound Phonebook to facilitate completion of the call would be

0182263742	for calls within the office at Tavistock
01822	for calls to the Tavistock local calling area (PSTN).

Call Event Sequence

1. Caller in Inner London dials 901822637424.
2. Inner London voip removes "9" .
3. Inner London voip passes remaining string, 01822637424on to the Tavistock voip
   at IP address 200.2.9.5.
4. The dialed string matches an inbound phonebook entry at the Tavistock voip, namely 0182263742.
5. The Tavistock voip rings one of the three FXS ports connected to the Tavistock
   key phone system.
6. The call will be routed to extension 7424 either by a human receptionist/
   operator or to an auto-attendant (which allows the caller to specify the
   extension to which they wish to be connected).

Site F calling Site D

A voip call from a Tavistock key extension to extension 3117 on the PBX in Inner London.

A. The required entry in the Tavistock Outbound Phonebook to facilitate origination of the call, would be “3”. The string 02073988 is added, preceding the “3”. The call would be directed to the Inner London voip’s IP address, 200.2.9.9.

B. The corresponding entry in the Inner-London Inbound Phonebook to facilitate completion of the call would be 020739883.

1. The caller in Tavistock picks up the phone receiver, presses a button on the key phone set. This button has been assigned to a particular voip channel.
2. The caller in Tavistock hears dial tone from the Tavistock voip.
3. The caller in Tavistock dials 02073983117.
4. The Tavistock voip sends the entire dialed string to the Inner-London voip
   at IP address 200.2.9.9.
5. The Inner-London voip matches the called digits 02073983117to its Inbound Phonebook entry “020739883, ” which it removes. Then it adds back the “3” as a prefix.
6. The Inner-London PBX dials extension 3117 in the office in Inner London.

Variations in PBX Characteristics

The exact dialing strings needed in the Outbound and Inbound Phonebooks of the MVP3010 will depend on the capabilities of the PBX. Some PBXs require trunk access codes (like an “8” or “9” to access an outside line or to access the VOIP network). Other PBXs can automatically distinguish between intra-PBX calls, PSTN calls, and VOIP calls.

Some PBX units can also insert digits automatically when they receive certain dialing strings from a phone station. For example, a PBX may be programmable to insert automatically the three-digit VOIP identifier strings into calls to be directed to analog VOIPs.

The MVP3010 offers complete flexibility for inter-operation with PBX units so that a coherent dialing scheme can be established to connect a company’s multiple sites together in a way that is convenient and intuitive for phone users. When working together with modern PBX units, the presence of the MVP3010 can be completely transparent to phone users within the company.

International Telephony Numbering Plan Resources

Due to the expansion of telephone number capacity to accommodate pagers, fax machines, wireless telephony, and other new phone technologies, numbering plans have been changing worldwide. Many new area codes have been established; new service categories have been established (for example, to accommodate GSM, personal numbering, corporate numbering, etc.). Below we list several web sites that present up-to-date information on the telephony numbering plans used around the world. While we find these to be generally good resources, we would note that URLs may change or become nonfunctional, and we cannot guarantee the quality of information on these sites.

URL	Description
http://phonebooth.interocitor.net /wtng	The World Telephone Numbering Guide presents excellent international numbering info that is both broad and detailed. This includes info on re-numbering plans carried out worldwide in recent years to accommodate new technologies.
http://www.oftel.gov.uk/numbers /number.htm	UK numbering plan from the Office of Telecommunications, the UK telephony authority.
http://www.itu.int/home/index.html	The International Telecommunications Union is an excellent source and authority on international telecom regulations and standards. National and international number plans are listed on this site.

URL	Description
http://kropla.com/phones.htm	Guide to international use of modems.
http://www.numberplan.org/	National and international numbering plans based on direct input from regulators worldwide. Includes lists of telecom carriers per country.
http://www.eto.dk/	European Telecommunications Office. Primarily concerned with mobile/wireless radiotelephony, GSM, etc.
http://www.eto.dk/ETNS.htm	European Telephony Numbering Space. Resources for pan-European telephony services, standards, etc. Part of ETO site.
http://www.regtp.de/en/reg_tele/start/fs_05.html	List of European telecom regulatory agencies by country (from German telecom authority).

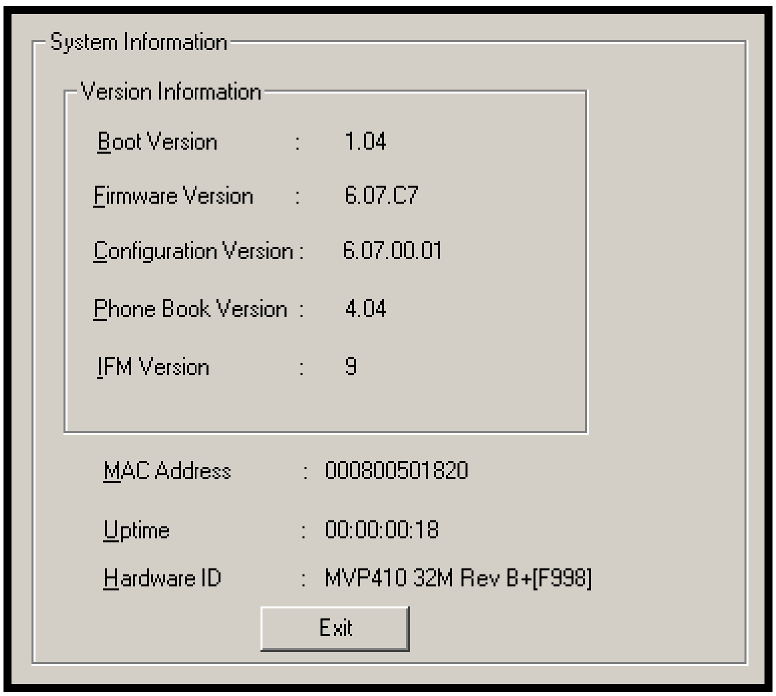
Chapter 8: Operation and Maintenance

Operation and Maintenance

Although most Operation and Maintenance functions of the software are in the **Statistics** group of screens, an important summary appears in the **System Information** of the **Configuration** screen group.

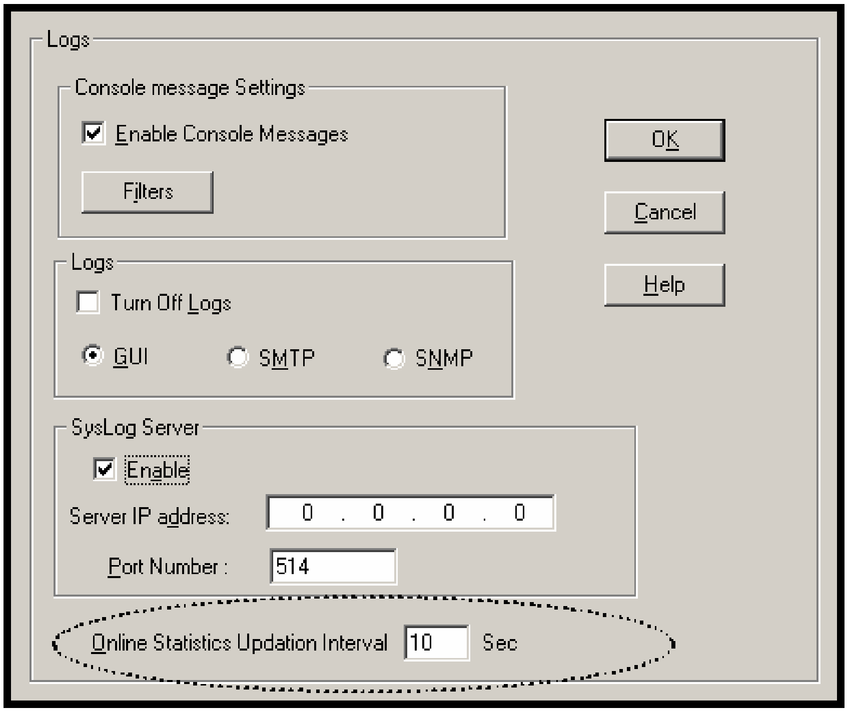
System Information screen

This screen presents vital system information at a glance. Its primary use is in troubleshooting. This screen is accessible via the **Configuration** pulldown menu, the **Configuration** sidebar menu, or by the keyboard shortcut **Ctrl + Alt + Y**.



System Information Parameter Definitions		
Field Name	**Values**	**Description**
Boot Version	nn.nn alpha-numeric	Indicates the version of the code that is used at the startup (booting) of the voip. The boot code version is independent of the software version.
Firmware Version	nn.nn.nn alpha-numeric	Indicates the version of the MultiVOIP firmware.
Configur-ation Version	nn.nn. nn.nn alpha-numeric	Indicates the version of the MultiVOIP configuration software.
Phone Book Version	nn.nn alpha-numeric	Indicates the version of the MultiVOIP phone book being used.
IFM Version	nn alpha-numeric	Indicates version of the IFM module, the device that performs the transformation between telephony signals and IP signals.
Mac Address	numeric	Denotes the number assigned as the voip unit's unique Ethernet address.
Up Time	days: hours: mm:ss	Indicates how long the voip has been running since its last booting.
Hardware ID	alpha-numeric	Indicates version of the MultiVOIP circuit board assembly being used.

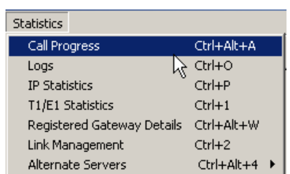
The frequency with which the System Information screen is updated is determined by a setting in the Logs screen



Statistics Screens

Ongoing operation of the MultiVOIP, whether it is in a MultiVOIP/PBX setting or MultiVOIP/telco-office setting, can be monitored for performance using the Statistics functions of the MultiVOIP software.

About Call Progress

<table>
<tr><th colspan="2">Accessing Call-Progress Statistics</th></tr>
<tr><td colspan="2">Channel Icons (Main Screen Lower Left)</td></tr>
<tr><td colspan="2">Channel icons are green when data traffic is present, red when idle. </td></tr>
<tr><td colspan="2">In the web GUI, call progress details can be viewed by clicking on an icon (one for each channel) arranged similarly on the web-browser screen.</td></tr>
<tr><td></td><td></td></tr>
<tr><td>Pulldown</td><td>Icon</td></tr>
<tr><td>
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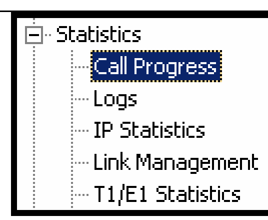
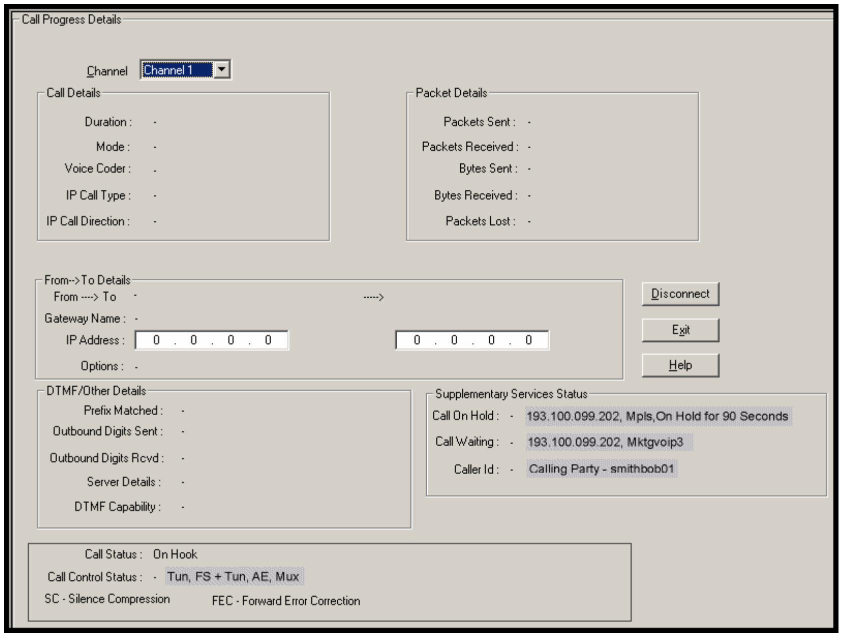
<tr><td>Shortcut</td><td>Sidebar</td></tr>
<tr><td><code>Ctrl + Alt + A</code></td><td>
</td></tr>
</table>

The Call Progress Details Screen



Call Progress Details: Field Definitions		
Field Name	**Values**	**Description**
Channel	1-n	Number of data channel or time slot on which the call is carried. This is the channel for which call-progress details are being viewed.
Call Details		
Duration	Hours: Minutes: Seconds	The length of the call in hours, minutes, and seconds (hh:mm:ss).
Mode	Voice or FAX	Indicates whether the call being described was a voice call or a FAX call.
Voice Coder	G.723, G.729, G.711, etc.	The voice coder being used on this call.
IP Call Type	H.323, SIP, or SPP	Indicates the Call Signaling protocol used for the call (H.323, SIP, or SPP).
IP Call Direction	incoming, outgoing	Indicates whether the call in question is an incoming call or an outgoing call.

Call Progress Details: Field Definitions		
Field Name	**Values**	**Description**
Packet Details		
Packets Sent	integer value	The number of data packets sent over the IP network in the course of this call.
Packets Rcvd	integer value	The number of data packets received over the IP network in the course of this call.
Bytes Sent	integer value	The number of bytes of data sent over the IP network in the course of this call.
Bytes Rcvd	integer value	The number of bytes of data received over the IP network in the course of this call.
Packets Lost	integer value	The number of voice packets from this call that were lost after being received from the IP network.

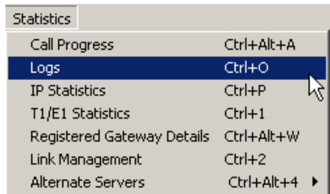
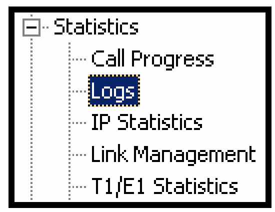
Call Progress Details: Field Definitions (cont'd)		
From – To Details		**Description**
Gateway Name (from)	alphanumeric string	Identifier for the VOIP gateway that handled the origination of this call.
IP Address (from)	x.x.x.x, where x has a range of 0 to 255	IP address from which the call was received.
Options	SC, FEC	Displays VOIP transmission options in use on the current call. These may include Forward Error Correction or Silence Compression.
Gateway Name (to)	alphanumeric string	Identifier for the VOIP gateway that handled the completion of this call.
IP Address (to)	x.x.x.x, where x has a range of 0 to 255	IP address to which the call was sent.
Options	SC, FEC	Displays VOIP transmission options in use on the current call. These may include Forward Error Correction or Silence Compression.

Call Progress Details: Field Definitions (cont'd)		
DTMF/Other Details		
Field Name	**Values**	**Description**
Prefix Matched	specified dialing digits	Displays the dialed digits that were matched to a phonebook entry.
Outbound Digits Sent	0-9, #, *	The digits transmitted by the MultiVOIP to the PBX/telco for this call.
Outbound Digits Received	0-9, #, *	Of the digits transmitted by the MultiVOIP to the PBX/telco for this call, these are the digits that were confirmed as being received.
Server Details	n.n.n.n (for n=0-255) and/or other server IP-related descriptions	The IP address (etc.) of the traffic control server (if any) being used (whether an H.323 gatekeeper, a SIP proxy, or an SPP registrar gateway) will be displayed here if the call is handled through that server.
DTMF Capability	inband, out of band Expressions differ slightly for different Call Signaling protocols (H.323, SIP, or SPP).	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." The corresponding field values differ for the 3 different voip protocols. For H.323, this field can display "Out of Band" or "Inband". For SIP it can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition. For SPP it can display "Out of Band RFC2833" or "Inband".

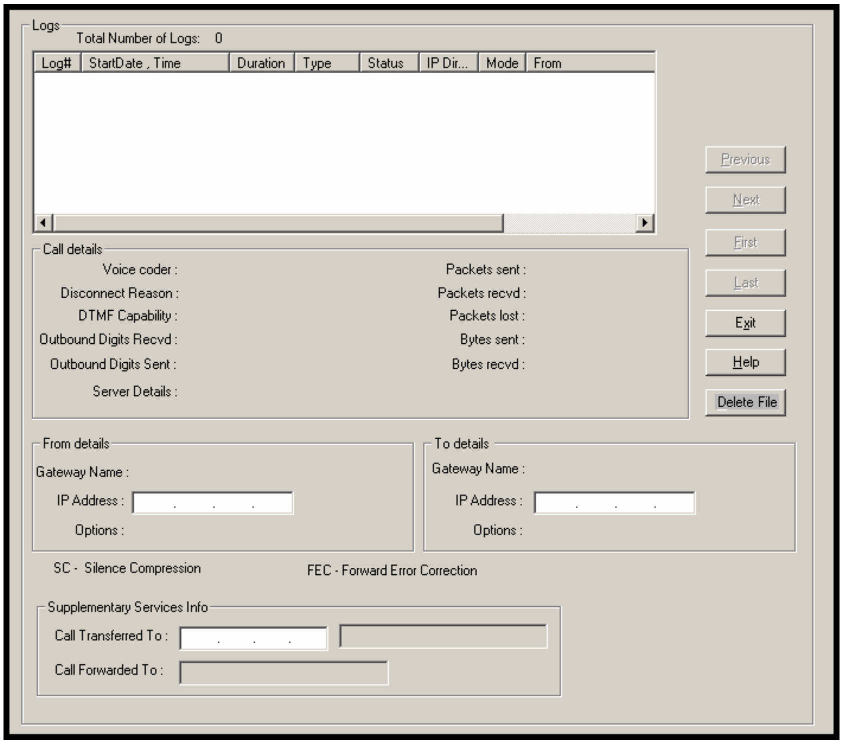
Call Progress Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Supplementary Services Status		
Call on Hold	alphanumeric	Describes held call by its IP address source, location/gateway identifier, and hold duration. Location/gateway identifiers comes from **Gateway Name** field in **Phone Book Configuration** screen of remote voip.
Call Waiting	alphanumeric	Describes waiting call by its IP address source, location/gateway identifier, and hold duration. Location/gateway identifiers comes from **Gateway Name** field in **Phone Book Configuration** screen of remote voip.
Caller ID	There are four values: "Calling Party + *identifier*"; "Alerting Party + *identifier*"; "Busy Party + *identifier*"; and "Connected Party + *identifier*"	This field shows the identifier and status of a remote voip (which has Call Name Identification enabled) with which this voip unit is currently engaged in some voip transmission. The status of the engagement (Connected, Alerting, Busy, or Calling) is followed by the identifier of a specific channel of a remote voip unit. This identifier comes from the "Caller Id" field in the **Supplementary Services** screen of the remote voip unit.

Call Progress Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Status fields		
Call Status	hangup, active	Shows condition of current call.
Call Control Status	Tun, FS + Tun, AE, Mux	Displays the H.323 version 4 features in use for the selected call. These include tunneling (Tun), Fast Start with tunneling (FS + Tun), Annex E multiplexed UDP call signaling transport (AE), and Q.931 Multiplexing (Mux). See **Phonebook Configuration Parameters** (in T1 or E1 chapters) for more on H.323v4 features.
Silence Compression	SC	"SC" stands for Silence Compression. With **Silence Compression** enabled, the MultiVOIP will not transmit voice packets when silence is detected, thereby reducing the amount of network bandwidth that is being used by the voice channel.
Forward Error Correction	FEC	"FEC" stands for Forward Error Correction. **Forward Error Correction** enables some of the voice packets that were corrupted or lost to be recovered. FEC adds an additional 50% overhead to the total network bandwidth consumed by the voice channel. Default = Off

About Logs

Accessing “Statistics: Logs”	
Pulldown	**Icon**
	
Shortcut	**Sidebar**
`Ctrl + O`	

The Logs Screen



Logs Screen Details: Field Definitions		
Field Name	**Values**	**Description**
Log # column	1 or higher	All calls are assigned an event number in chronological order, with the most recent call having the highest event number.
Start Date,Time column	dd:mm:yyyy hh:mm:ss	The starting time of the call (event). The date is presented as a day expression of one or two digits, a month expression of one or two digits, and a four-digit year. This is followed by a time-of-day expression presented as a two-digit hour, a two-digit minute, and a two-digit seconds value. (statistics, logs) field
Duration column	hh:mm:ss	This describes how long the call (event) lasted in hours, minutes, and seconds.
Type	H.323, SIP, or SPP	Indicates the Call Signaling protocol used for the call (H.323, SIP, or SPP).
Status column	success or failure	Displays the status of the call, i.e., whether the call was completed successfully or not.
IP Direction	incoming, outgoing	Indicates whether the call is "incoming" or "outgoing" with respect to the gateway.
Mode column	voice or FAX	Indicates whether the (event) being described was a voice call or a FAX call.
From column	gateway name	Displays the name of the voice gateway that originates the call.
To column	gateway name	Displays the name of the voice gateway that completes the call.
Special Buttons		
Previous	--	Displays log entry before currently selected one.
Next	--	Displays log entry after currently selected one.
First	--	Displays first log entry
Last	--	Displays last log entry.
Delete File	--	Deletes selected log file.

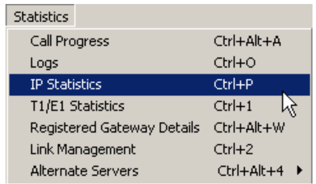
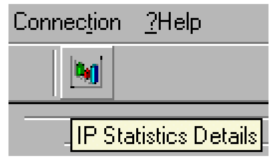
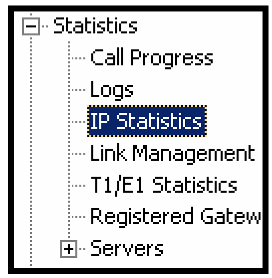
Logs Screen Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Details		
Voice coder	G.723, G.729, G.711, etc.	The voice coder being used on this call.
Disconnect Reason	Values are "Normal" and "Local" disconnection.	Indicates whether the call was disconnected simply because the desired conversation was done or some other irregular cause occasioned disconnection (e.g., a technical error or failure).
DTMF Capability	inband, out of band Expressions differ slightly for different Call Signaling protocols (H.323, SIP, or SPP).	Indicates whether the DTMF dialing digits are carried "Inband" or "Out of Band." The corresponding field values differ for the 3 different voip protocols. For H.323, this field can display "Out of Band" or "Inband". For SIP it can display either "Out of Band RFC2833" or "Out of Band SIP INFO" to indicate the out-of-band condition or "Inband" to indicate the in-band condition. For SPP it can display "Out of Band RFC2833" or "Inband".
Outbound Digits Received	0-9, #, *	The digits, sent by MultiVOIP to PBX/telco, that were acknowledged as having been received by the remote voip gateway.
Outbound Digits Sent	0-9, #, *	The digits transmitted by the MultiVOIP to the PBX/telco for this call.

Logs Screen Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Details		
Server Details	n.n.n.n for n= 0-255	When the MultiVOIP is operating in the non-direct mode (with Gatekeeper in H.323 mode; with proxy in SIP mode; or in the client/server configuration of SPP mode), this field shows the IP address of the server that is directing IP phone traffic.
Packets sent	integer value	The number of data packets sent over the IP network in the course of this call.
Packets received	integer value	The number of data packets received over the IP network in the course of this call.
Packets loss (lost)	integer value	The number of voice packets from this call that were lost after being received from the IP network.
Bytes sent	integer value	The number of bytes of data sent over the IP network in the course of this call.
Bytes received	integer value	The number of bytes of data received over the IP network in the course of this call.

Logs Screen Details: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Call Details (cont'd)		
FROM Details		
Gateway Name	alphanumeric string	Identifier for the VOIP gateway that originated this call.
IP Address	x.x.x.x, where x has a range of 0 to 255	IP address of the VOIP gateway from which the call was received.
Options	FEC, SC	Displays VOIP transmission options used by the VOIP gateway originating the call. These may include Forward Error Correction or Silence Compression.
TO Details		
Gateway Name	alphanumeric string	Identifier for the VOIP gateway that completed (terminated) this call.
IP Address	x.x.x.x, where x has a range of 0 to 255	IP address of the VOIP gateway at which the call was completed (terminated).
Options		Displays VOIP transmission options used by the VOIP gateway terminating the call. These may include Forward Error Correction or Silence Compression.

Logs Screen Details: Field Definitions (cont'd)		
Supplementary Services Info		
Call Transferred To	phone number string	Number of party called in transfer.
Call Forwarded To	phone number string	Number of party called in forwarding.

About IP Statistics

<table>
<tr><th colspan="2">Accessing IP Statistics</th></tr>
<tr><th>Pulldown</th><th>Icon</th></tr>
<tr><td>
</td><td>
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td><code>Ctrl + P</code></td><td>
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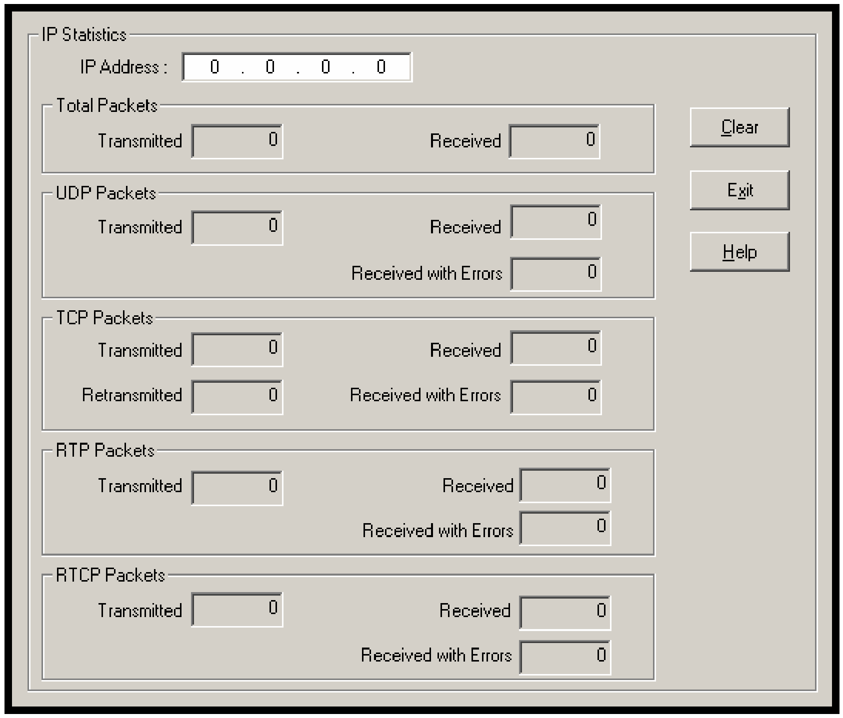
IP Statistics Screen



IP Statistics: Field Definitions		
Field Name	**Values**	**Description**
		UDP versus TCP. (User Datagram Protocol versus Transmission Control Protocol). UDP provides unguaranteed, connectionless transmission of data across an IP network. By contrast, TCP provides reliable, connection-oriented transmission of data.

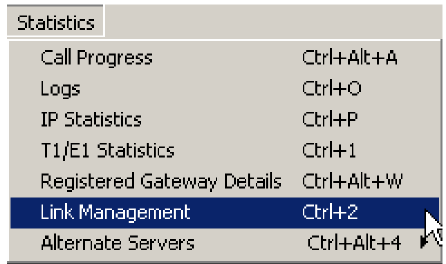
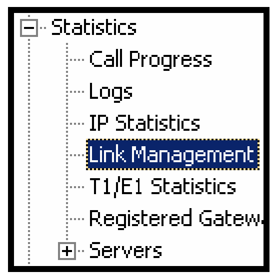
IP Statistics: Field Definitions		
Field Name	**Values**	**Description**
		UDP versus TCP (continued). Both TCP and UDP split data into packets called "datagrams." However, TCP includes extra headers in the datagram to enable retransmission of lost packets and reassembly of packets into their correct order if they arrive out of order. UDP does not provide this. Lost UDP packets are unretrievable; that is, out-of-order UDP packets cannot be reconstituted in their proper order.. Despite these obvious disadvantages, UDP packets can be transmitted much faster than TCP packets -- as much as three times faster. In certain applications, like audio and video data transmission, the need for high speed outweighs the need for verified data integrity. Sound or pictures often remain intelligible despite a certain amount of lost or disordered data packets (which appear as static).
IP Address	n.n.n.n 0 - 255	IP address of the MultiVOIP. For an IP address to be displayed here, the MultiVOIP must have DHCP enabled. Its IP address, in such a case, is assigned by the DHCP server.
"Clear" button	--	Clears packet tallies from memory.
Total Packets		Sum of data packets of all types.
Transmitted	integer value	Total number of packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Total number of packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.

IP Statistics: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Total Packets (cont'd)		Sum of data packets of all types.
Received with Errors	integer value	Total number of error-laden packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
UDP Packets		User Datagram Protocol packets.
Transmitted	integer value	Number of UDP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of UDP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden UDP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
TCP Packets		Transmission Control Protocol packets.
Transmitted	integer value	Number of TCP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of TCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden TCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.

IP Statistics: Field Definitions (cont'd)		
RTP Packets		Voice signals are transmitted in Realtime Transport Protocol packets. RTP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of RTP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden RTP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
RTCP Packets		Realtime Transport Control Protocol packets convey control information to assist in the transmission of RTP (voice) packets. RTCP packets are a type or subset of UDP packets.
Transmitted	integer value	Number of RTCP packets transmitted by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received	integer value	Number of RTCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.
Received with Errors	integer value	Number of error-laden RTCP packets received by this VOIP gateway since the last "clearing" or resetting of the counter within the MultiVOIP software.

About Link Management

The Link Management screen is essentially an automated utility for pinging endpoints on your voip network. This utility generates pings of variable sizes at variable intervals and records the response to the pings.

<table>
<tr><th colspan="2">Accessing Link Management</th></tr>
<tr><th colspan="2">Pulldown</th></tr>
<tr><td colspan="2">
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut // Icon</th><th>Sidebar</th></tr>
<tr><td>Ctrl + 2 // none</td><td>
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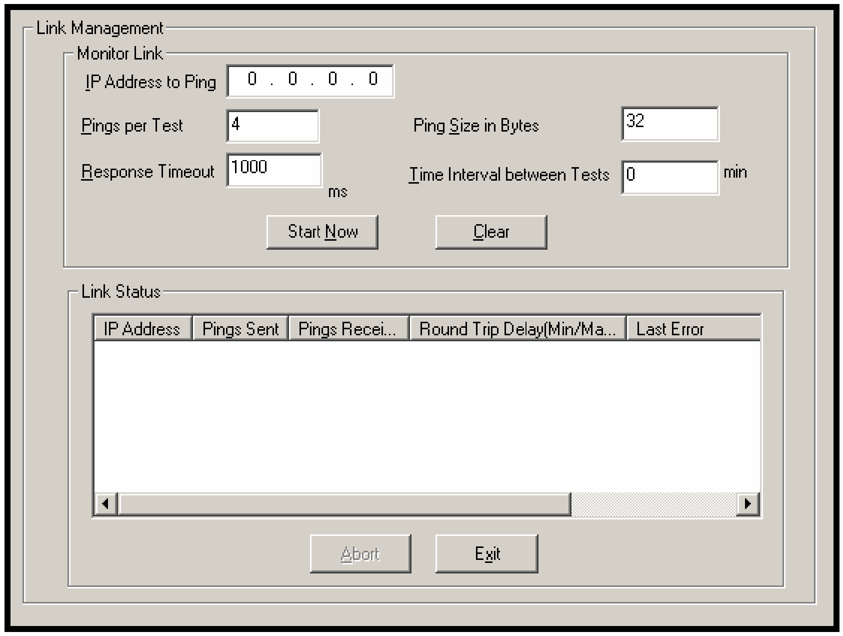
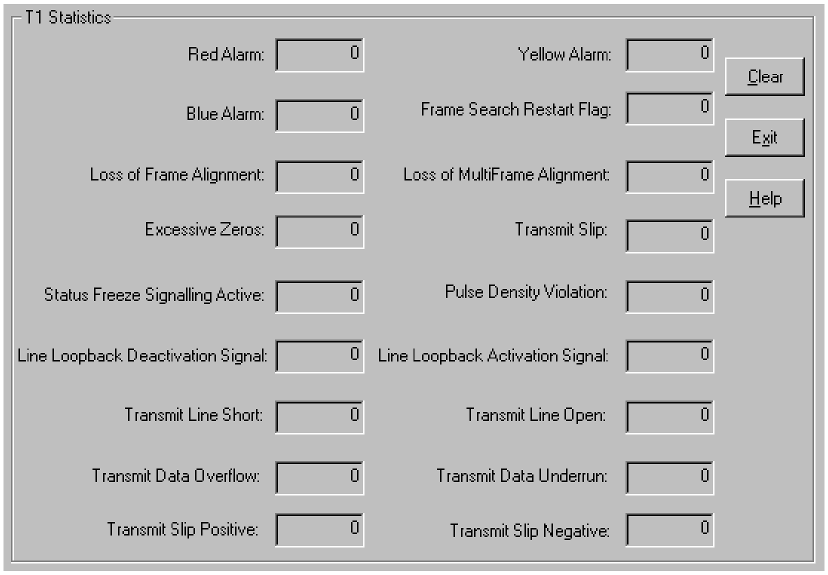

Link Management
Monitor Link
IP Address to Ping
0 . 0 . 0 . 0
Pings per Test
4
Ping Size in Bytes
32
Response Timeout
1000
ms
Time Interval between Tests
0
min
Start Now
Clear
Link Status
IP Address
Pings Sent
Pings Recei...
Round Trip Delay(Min/Ma...
Last Error
Abort
Exit

Link Management screen Field Definitions		
Field Name	**Values**	**Description**
Monitor Link fields		
IP Address to Ping	a.b.c.d 0-255	This is the IP address of the target endpoint to be pinged.
Pings per Test	1-999	This field determines how many pings will be generated by the Start Now command.
Response Timeout	500 – 5000 milliseconds	The duration after which a ping will be considered to have failed.
Ping Size in Bytes	32 – 128 bytes	This field determines how long or large the ping will be.
Timer Interval between Pings	0 or 30 – 6000 minutes	This field determines how long of a wait there is between one ping and the next.
Start Now command button	--	Initiates pinging.
Clear command button	--	Erases ping parameters in Monitor Link field group and restores default values.

Link Management screen Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Link Status Parameters		These fields summarize the results of pinging.
IP Address column	a.b.c.d 0-255	Target of ping.
No. of Pings Sent	as listed	Number of pings sent to target endpoint.
No. of Pings Received	as listed	Number of pings received by target endpoint.
Round Trip Delay (Min/Max/ Avg)	as listed, in milliseconds	Displays how long it took from time ping was sent to time ping response was received.
Last Error	as listed	Indicates when last data error occurred.

T1 Statistics Screen

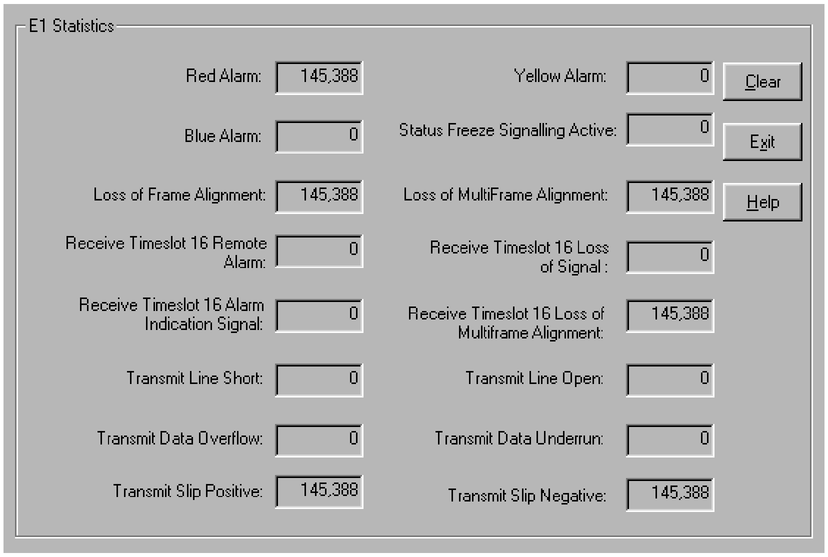


T1 Statistics: Field Definitions		
Field Name	**Values**	**Description**
Red Alarm	Integer tally of alarms counted since last reset.	The alarm condition declared when a device receives no signal or cannot synchronize to the signal being received. A Red Alarm is generated if the incoming data stream has no transitions for 176 consecutive pulse positions.
Blue Alarm	Tally since last reset.	Alarm signal consisting of all 1's (including framing bit positions) which indicates disconnection or failure of attached equipment.
Loss of Frame Alignment	Tally since last reset.	Loss of data frame synchronization.
Excessive Zeroes	Tally since last reset.	Displayed value will increment if consecutive zeroes beyond a set threshold are detected. I.e., tally increments if more than 7 consecutive zeroes in the received data stream are detected under B8ZS line coding, or if 15 consecutive zeroes are detected under AMI line coding.
Status Freeze Signaling Active		Signaling has been frozen at the most recent values due to loss of frame alignment, loss of multiframe alignment or due to a receive slip.
Line Loopback Deactivation Signal		Line loopback deactivation signal has been detected in the receive bit stream.
Transmit Line Short		A short exists between the transmit pair for at least 32 consecutive pulses.
Transmit Data Overflow		For use by MTS Technical Support personnel.
Transmit Slip Positive		The frequency of the transmit clock is less than the frequency of the transmit system interface working clock. A frame is repeated.

T1 Statistics: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Yellow Alarm	Tally since last reset.	The alarm signal sent by a remote T1/E1 device to indicate that it sees no receive signal or cannot synchronize on the receive signal.
Frame Search Restart Flag		[To be supplied.]
Loss of MultiFrame Alignment	Tally since last reset.	In D4 or ESF mode, displayed value will increment if multiframe alignment has been lost or if loss of frame alignment has been detected.
Transmit Slip	Tally since last reset.	Slip in transmitted data stream. Slips indicate a clocking mismatch (or lack of synchronization) between T1/E1 devices. When slips occur, data may be lost or repeated.
Pulse Density Violation		The pulse density of the received data stream is below the requirement defined by ANSI T1.403 or more than 15 consecutive zeros are detected.
Line Loopback Activation Signal		The line loopback activation signal has been detected in the received bit stream.
Transmit Line Open		At least 32 consecutive zeros were transmitted.
Transmit Data Underrun		For use by MTS Technical Support Personnel.
Transmit Slip Negative		The frequency of the transmit clock is greater than the frequency of the transmit system interface working clock. A frame is skipped.

T1 Statistics: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Bipolar Violation	Integer tally of violation count since last reset.	Two successive pulses of the same polarity have been received and these pulses are not part of zero substitution. On an AMI-encoded line, this represents a line error. On a B8ZS line, this may represent the substitution for a string of 8 zeroes.
Receive Slip	Tally since last reset.	A receive slip (positive or negative) has occurred. Slips indicate a clocking mismatch (or lack of synchronization) between T1/E1 devices. When slips occur, data may be lost or repeated.

E1 Statistics Screen



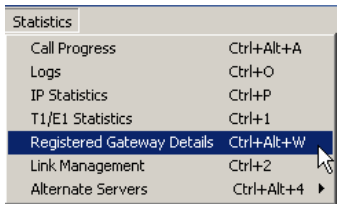
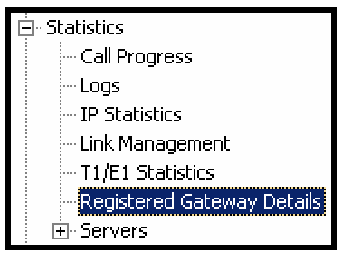
E1 Statistics: Field Definitions		
Field Name	**Values**	**Description**
Red Alarm	Integer tally of alarms counted since last reset.	The alarm condition declared when a device receives no signal or cannot synchronize to the signal being received. A Red Alarm is generated if the incoming data stream has no transitions for 176 consecutive pulse positions.
Blue Alarm	Tally since last reset.	Alarm signal consisting of all 1's (including framing bit positions) which indicates disconnection or failure of attached equipment.
Loss of Frame Alignment	Tally since last reset.	Loss of data frame synchronization.

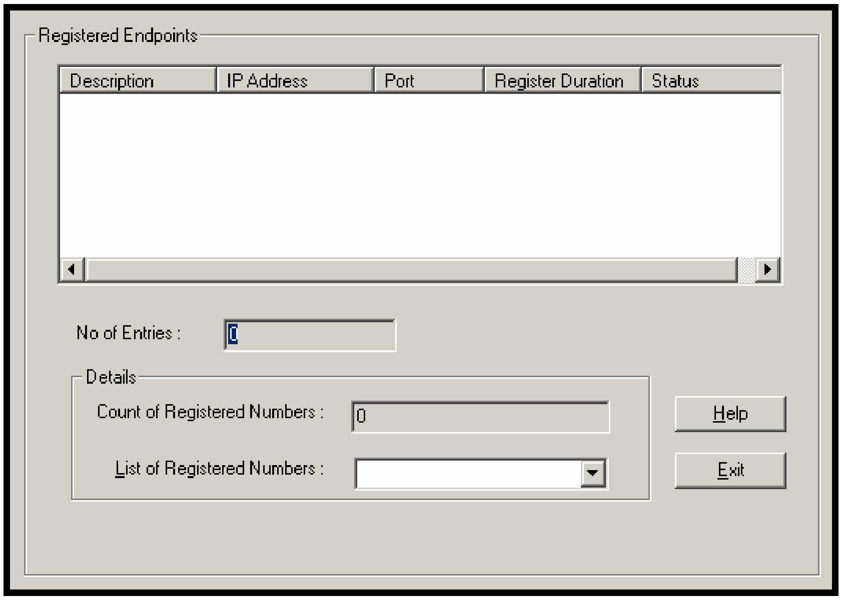
E1 Statistics: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Receive Timeslot 16 Alarm Indication Signal		Detected alarm indication signal in timeslot 16 according to ITU-T G.775. Indicates the incoming time slot 16 contains less than 4 zeros in each of two consecutive time slot 16 multiframe periods.
Transmit Line Short		A short exists between the transmit pair for at least 32 consecutive pulses.
Transmit Data Overflow		For use by MTS personnel.
Transmit Slip Positive		The frequency of the transmit clock is less than the frequency of the transmit system interface working clock. A frame is repeated.
Yellow Alarm	Tally since last reset.	The alarm signal sent by a remote T1/E1 device to indicate that it sees no receive signal or cannot synchronize on the receive signal.
Status Freeze Signaling Active		Signaling has been frozen at the most recent values due to loss of frame alignment, loss of multiframe alignment or due to a receive slip.
Loss of MultiFrame Alignment	Tally since last reset.	In D4 or ESF mode, displayed value will increment if multiframe alignment has been lost or if loss of frame alignment has been detected.
Receive Timeslot 16 Loss of Signal		The time slot 16 data stream contains all zeros for at least 16 contiguously received time slots.

E1 Statistics: Field Definitions (cont'd)		
Field Name	**Values**	**Description**
Receive Timeslot 16 Loss of MultiFrame Alignment		The framing pattern '0000' in 2 consecutive CAS multiframes were not found or in all time slot 16 of the previous multiframe all bits were reset.
Transmit Line Open		At least 32 consecutive zeroes were transmitted.
Transmit Data Underrun		For use by MTS Technical Support Personnel.
Transmit Slip Negative		The frequency of the transmit clock is greater than the frequency of the transmit system interface working clock. A frame is skipped.
Bipolar Violation	Integer tally of violation count since last reset.	Bipolar Violation (or BPV) refers to two successive pulses of the same polarity on the E1 line. On an AMI-encoded line, this represents a line error. On a B8ZS line, this may represent the substitution for a string of 8 zeroes.
Excessive Zeroes	Tally since last reset.	Displayed value will increment if consecutive zeroes beyond a set threshold are detected. I.e., tally increments if more than 7 consecutive zeroes in the received data stream are detected under B8ZS line coding, or if 15 consecutive zeroes are detected under AMI line coding.
Transmit Slip	Tally since last reset.	Slip in transmitted data stream. Slips indicate a clocking mismatch (or lack of synchronization) between T1/E1 devices. When slips occur, data may be lost or repeated.
Receive Slip	Tally since last reset.	Slip in received data stream. Slips indicate a clocking mismatch (or lack of synchronization) between T1/E1 devices. When slips occur, data may be lost or repeated.

About Registered Gateway Details

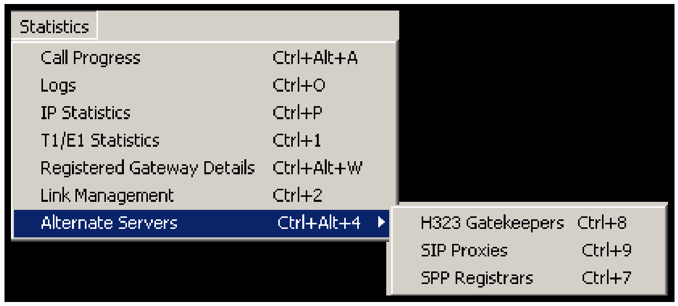
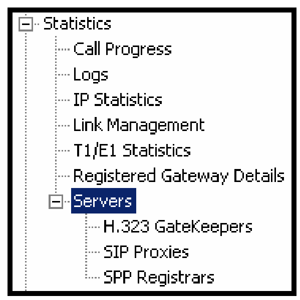
The Registered Gateway Details screen presents a real-time display of the special operating parameters of the Single Port Protocol (SPP). These are configured in the **Call Signaling** screen and in the **Add/Edit Outbound PhoneBook** screen.

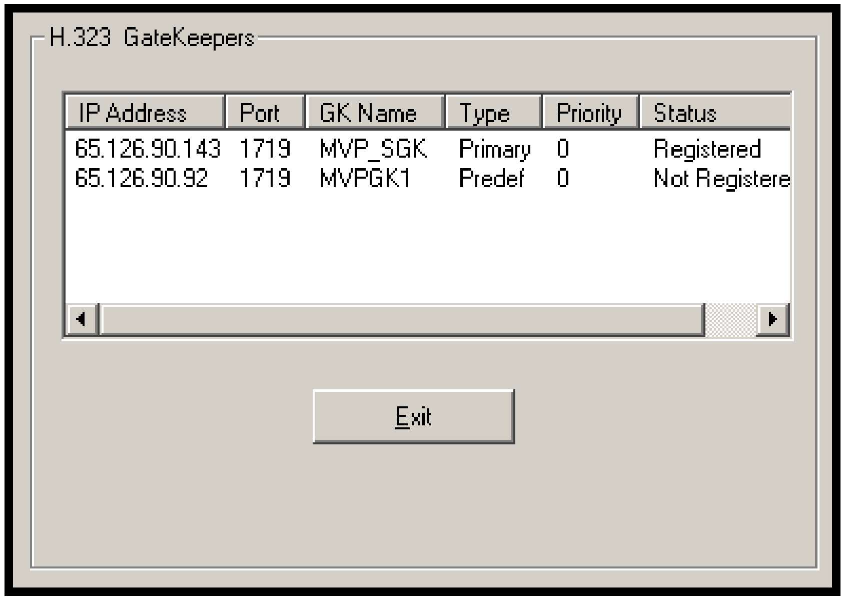
Accessing Registered Gateway Details	
Pulldown	**Shortcut**
	`Ctrl + Alt + W`
Sidebar	
	


Registered Endpoints
Description
IP Address
Port
Register Duration
Status
No of Entries :
0
Details
Count of Registered Numbers :
0
List of Registered Numbers :
Help
Exit

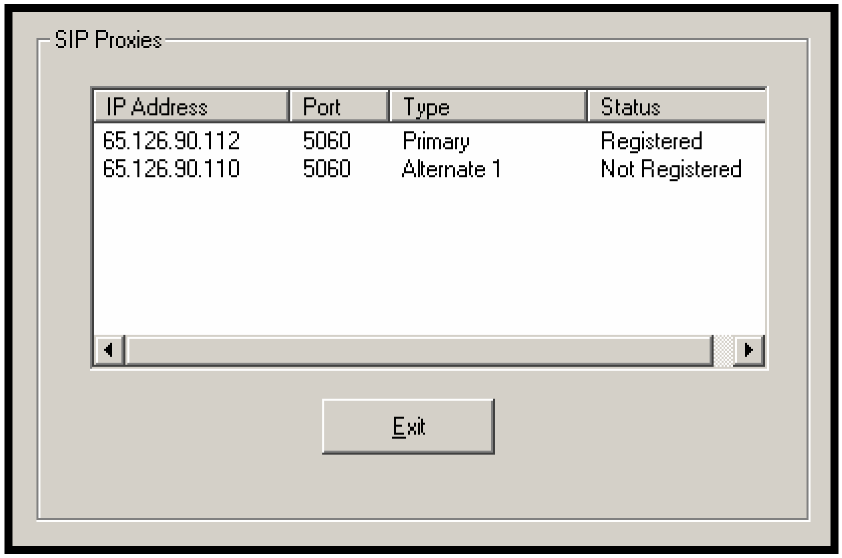
Registered Gateway Details: Field Definitions		
Field Name	**Values**	**Description**
Column Headings		
Description	alphanumeric	This is a descriptor for a particular voip gateway unit. This descriptor should generally identify the physical location of the unit (e.g., city, building, etc.) and perhaps even its location in an equipment rack.
IP Address	n.n.n.n, for n = 0-255	The RAS address for the gateway.
Port		Port by which the gateway exchanges H.225 RAS messages with the gatekeeper. .
Register Duration		The time remaining in seconds before the TimeToLive timer expires. If the gateway fails to reregister within this time, the endpoint is unregistered.
Status		The current status of the gateway, either registered or unregistered.
No. of Entries		The number of gateways currently registered to the Registrar. This includes all SPP clients registered and the Registrar itself.
Details		
Count of Registered Numbers		If a registered gateway is selected (by clicking on it in the screen), The "Count of Registered Numbers" will indicate the number of registered phone numbers for the selected gateway. When a client registers, all of its inbound phonebook's phone numbers become registered.
List of Registered Numbers		Lists all of the registered phone numbers for the selected gateway.

About Alternate Server Statistics

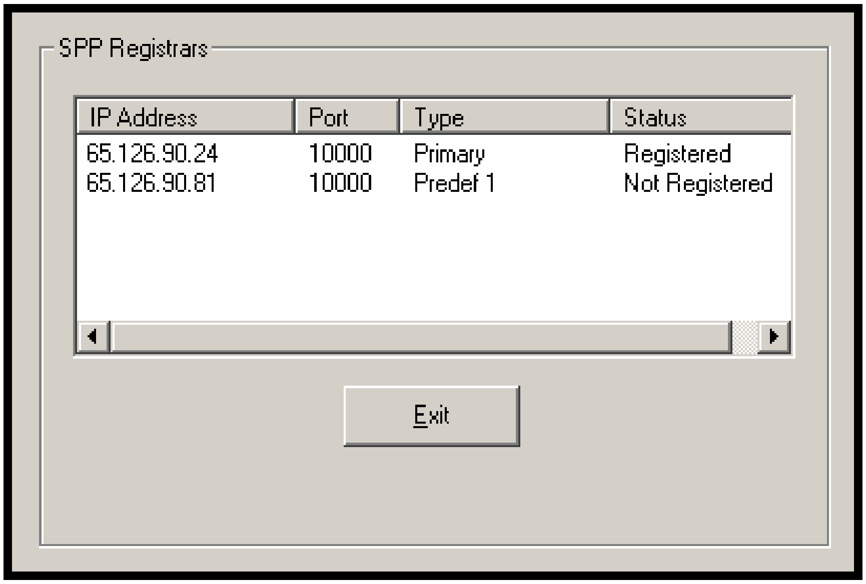
<table>
<tr><th colspan="2">Accessing Alternate Server Statistics</th></tr>
<tr><th colspan="2">Pulldown</th></tr>
<tr><td colspan="2">
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut</th><th>Sidebar</th></tr>
<tr><td><code>Ctrl + Alt + 4</code></td><td>
</td></tr>
</table>



H.323 Gatekeepers (Statistics, Servers): Field Definitions		
Field Name	**Values**	**Description**
Column Headings		
IP Address	n.n.n.n, for n = 0-255	The IP address of the gatekeeper.
Port		TDMA time slot used for communication between MultiVOIP unit and the gatekeeper that serves it.
GK Name	alpha-numeric string	Identifier for gatekeeper.
Type	Primary, Predefined	This field describes the type of gateway as which the MultiVOIP is defined with respect to the gatekeeper.
Priority		Priority refers to … .
Status	registered, not registered	The current status of the gateway, either registered or unregistered.



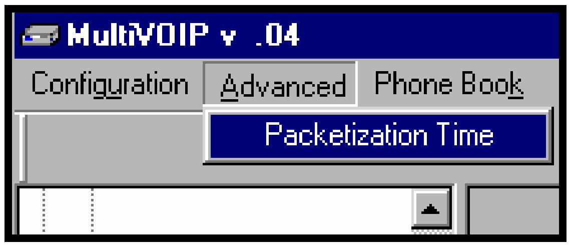
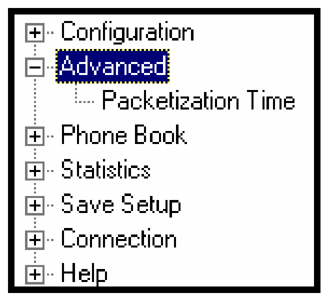
SIP Proxies (Statistics, Servers): Field Definitions		
Field Name	**Values**	**Description**
Column Headings		
IP Address	n.n.n.n, for n = 0-255	The IP address of the SIP proxy by which the MultiVOIP is governed.
Port		TDMA time slot used for communication between MultiVOIP unit and the SIP Proxy that governs it.
Type	Primary, Alternate	This field describes the type of gateway as which the MultiVOIP is defined with respect to the gatekeeper.
Status	registered, not registered	The current status of the MultiVOIP gateway with respect to the SIP proxy, either registered or unregistered.



SPP Registrars (Statistics, Servers): Field Definitions		
Field Name	**Values**	**Description**
Column Headings		
IP Address	n.n.n.n, for n = 0-255	The IP address of the gatekeeper.
Port		TDMA time slot used for communication between MultiVOIP unit and the gatekeeper that serves it.
Type	Primary, Predefined	This field describes the type of gateway as which the MultiVOIP is defined with respect to the gatekeeper.
Status	registered, not registered	The current status of the gateway, either registered or unregistered.

About Packetization Time

You can use the **Packetization Time** screen to specify definite packetization rates for coders selected in the Voice/FAX Parameters screen (in the "Coder Options" group of fields). The Packetization Time screen is accessible under the "Advanced" options entry in the sidebar list of the main voip software screen. In dealing with RTP parameters, the Packetization Time screen is closely related to both Voice/FAX Parameters and to IP Statistics. It is located in the "Advanced" group for ease of use.

<table>
<tr><th colspan="2">Accessing Packetization Time</th></tr>
<tr><th colspan="2">Pulldown</th></tr>
<tr><td colspan="2">
</td></tr>
<tr><td></td><td></td></tr>
<tr><th>Shortcut/Icon</th><th>Sidebar</th></tr>
<tr><td>none/none</td><td>
</td></tr>
</table>

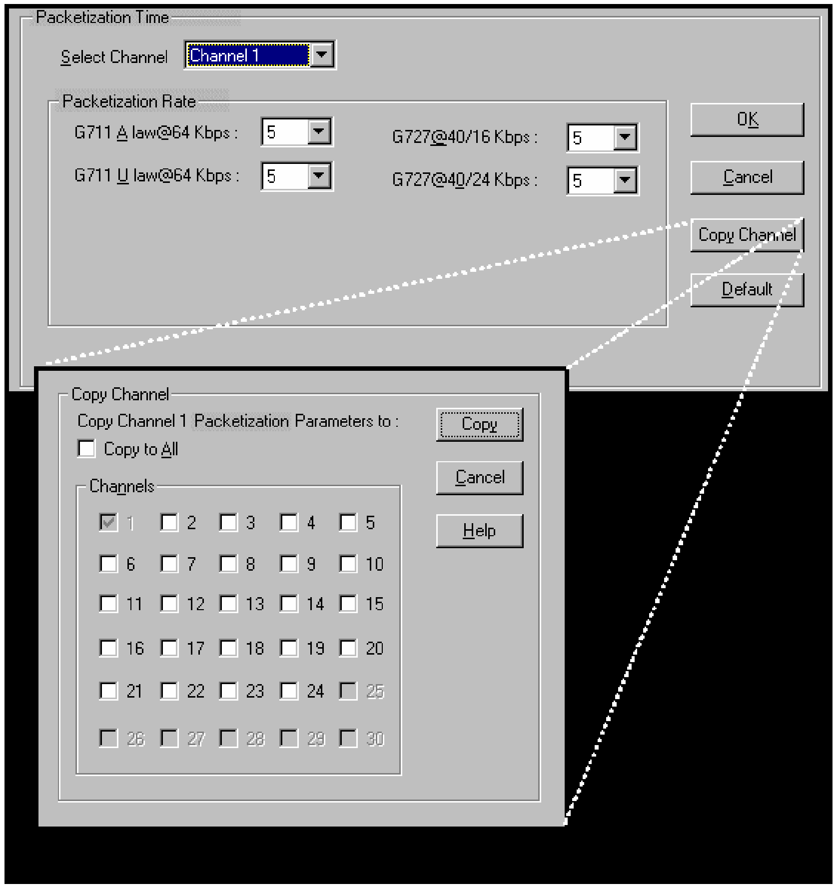
Packetization Time Screen

Packetization rates can be set separately for each channel.

The table below presents the ranges and increments for packetization rates.

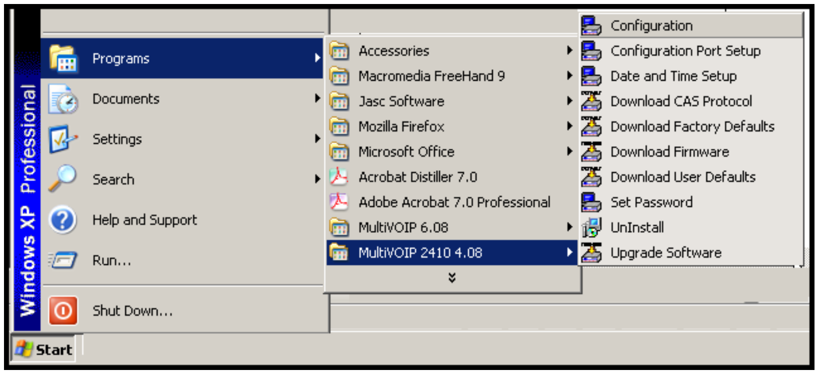
Packetization Ranges and Increments		
Coder Types	Range (in Kbps); {default value}	Increments (in Kbps)
G711, G726, G727	5-120 {5}	5
G723	30-120 {30}	30
G729	10-120 {10}	10
Netcoder	20-120 {20}	20

Once the packetization rate has been set for one channel, it can be copied into other channels.



MultiVoip Program Menu Items

After the MultiVoip program is installed on the PC, it can be launched from the **Programs** group of the Windows **Start** menu (**Start** | **Programs** | **MultiVOIP** ____ | …). In this section, we describe the software functions available on this menu.



Several basic software functions are accessible from the MultiVoip software menu, as shown below.

MultiVOIP Program Menu	
Menu Selection	**Description**
Configuration	Select this to enter the Configuration program where values for IP, telephony, and other parameters are set.
Configuration Port Setup	Select this to access the **COM Port Setup** screen of the MultiVOIP Configuration program.
Date and Time Setup	Select this for access to set calendar/clock used for data logging.

MultiVOIP Program Menu (cont'd)	
Menu Selection	**Description**
Download CAS Protocol	The CAS protocol code allows the VOIP to interact properly with the PBX or central-office switch that it serves. The need to download CAS protocols arises for only a small minority of VOIP users, and only when PBX/switch is found to be incompatible with standard protocols.
Download Factory Defaults	Select this to return the configuration parameters to the original factory values.
Download Firmware	Select this to download new versions of firmware as enhancements become available.
Download User Defaults	To be used after a full set of parameter values, values specified by the user, have been saved (using Save Setup). This command loads the saved user defaults into the MultiVOIP.
Set Password	Select this to create a password for access to the MultiVOIP software programs (**Program** group commands, Windows GUI, web browser GUI, & FTP server). Only the FTP Server function *requires* a password for access. The FTP Server function also requires that a username be established along with the password.
Uninstall	Select this to uninstall the MultiVOIP software (most, but not all components are removed from computer when this command is invoked).
Upgrade Software	Loads firmware (including H.323 stack) and settings from the controller PC to the MultiVOIP unit. User can choose whether to load Factory Default Settings or Current Configuration settings.

"Downloading" here refers to transferring program files from the PC to the nonvolatile "flash" memory of the MultiVOIP. Such transfers are made via the PC's serial port. This can be understood as a "download" from the perspective of the MultiVOIP unit.

When new versions of the MultiVoip software become available, they will be posted on MultiTech's web or FTP sites. Although transferring updated program files from the MultiTech web/FTP site to the user's PC can generally be considered a download (from the perspective of the PC), this type of download cannot be initiated from the MultiVoip software's Program menu command set.

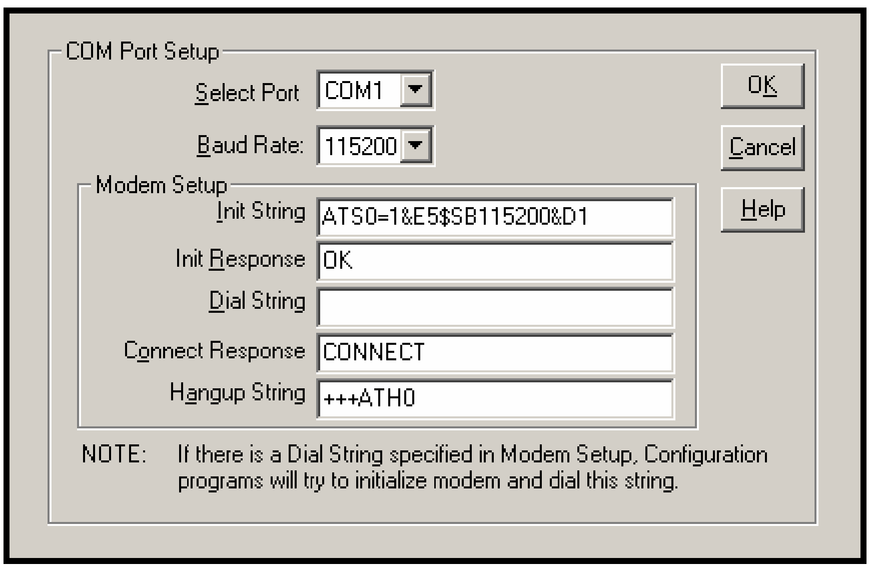
Generally, updated firmware must be downloaded from the MultiTech web/FTP site to the PC before it can be loaded from the PC to the MultiVOIP.

Configuration Option

The "Configuration" option in the MultiVOIP Program menu launches the MultiVOIP Configuration software program.

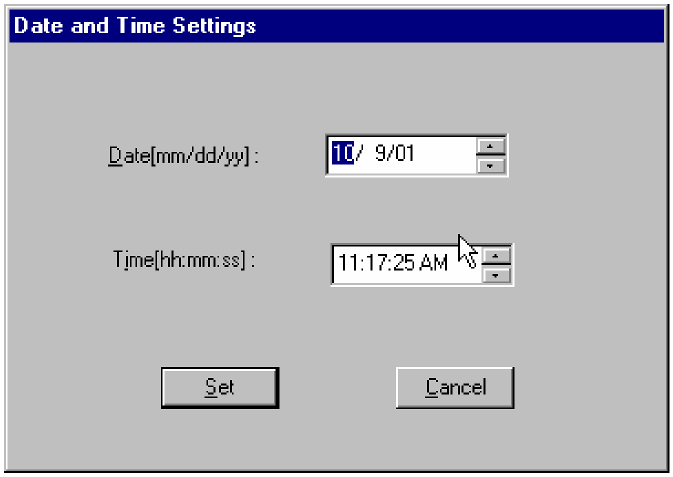
Configuration Port Setup

The Configuration Port Setup option in the MultiVOIP Program menu brings up the **COM Port Setup** screen of the MultiVOIP configuration software.



Date and Time Setup

The dialog box below allows you to set the time and date indicators of the MultiVOIP system.



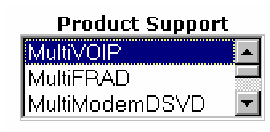
Obtaining Updated Firmware

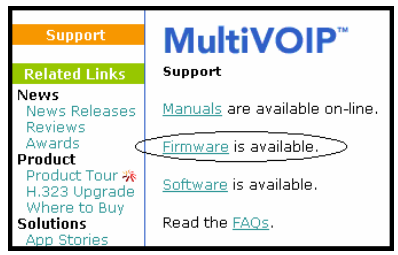
Generally, updated firmware must be downloaded from the MultiTech web/FTP site to the user's PC before it can be downloaded from that PC to the MultiVOIP.

Note that the structure of the MultiTech web/FTP site may change without notice. However, firmware updates can generally be found using standard web techniques. For example, you can access updated firmware by doing a search or by clicking on **Support**.

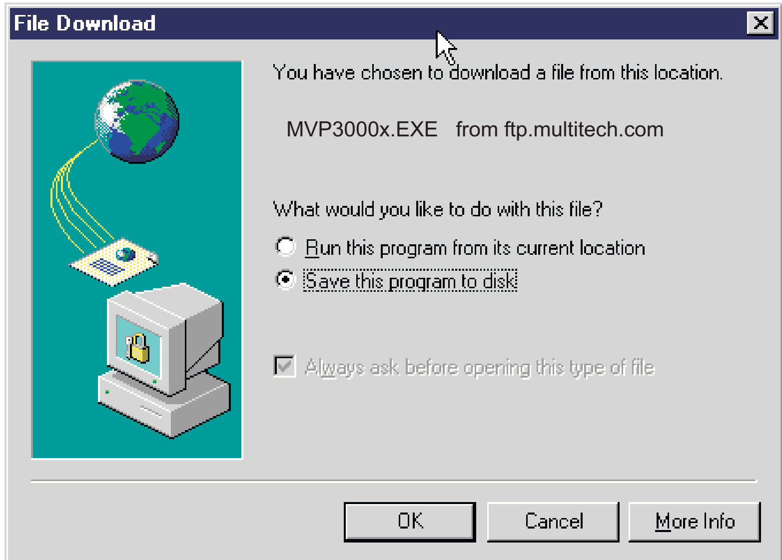
If you conduct a search, for example, on the word "MultiVoip," you will be directed to a list of firmware that can be downloaded.

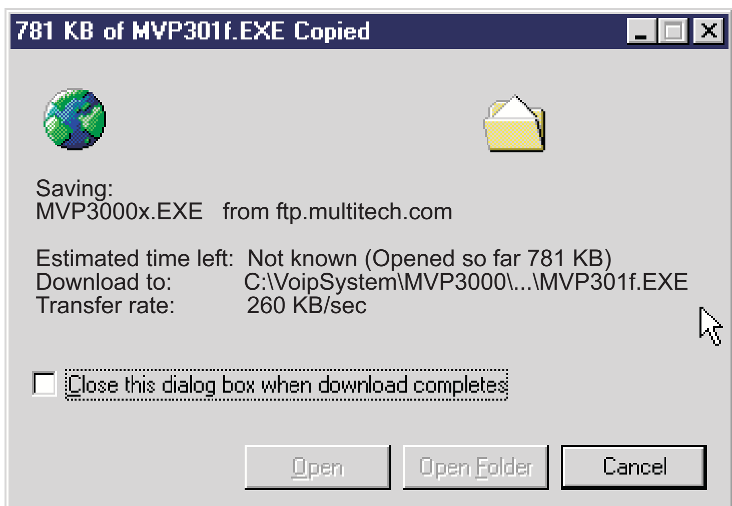
If you choose **Support**, you can select "MultiVoip" in the **Product Support** menu and then click on **Firmware** to find MultiVOIP resources.



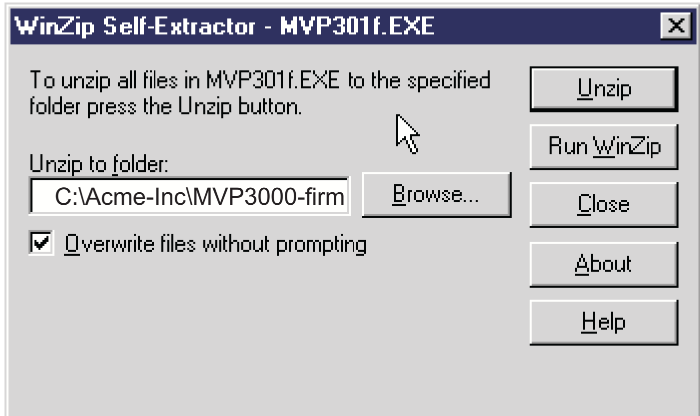


Once the updated firmware has been located, it can be downloaded from the web/ftp site using normal PC/Windows procedures. While the next 3 screens below pertain to the MVP3010, similar screens will appear for any MultiVOIP model described in this manual.





Generally, the firmware file will be a self-extracting compressed file (with .zip extension), which must be expanded (decompressed, or "unzipped") on the user's PC in a user-specified directory.



Implementing a Software Upgrade

MultiVOIP software can be upgraded locally using a single command at the MultiVOIP Windows GUI, namely **Upgrade Software**. This command downloads firmware (including the H.323 stack), and factory default settings from the controller PC to the MultiVOIP unit.

When using the MultiVOIP Windows GUI, firmware and factory default settings can also be transferred from controller PC to MultiVOIP piecemeal using separate commands.

When using the MultiVOIP web browser GUI to control/configure the voip remotely, upgrading of software must be done on a piecemeal basis using the FTP Server function of the MultiVOIP unit.

When performing a piecemeal software upgrade (whether from the Windows GUI or web browser GUI), follow these steps in order:

1. Identify Current Firmware Version
2. Download Firmware
3. Download Factory Defaults

When upgrading firmware, the software commands “Download Firmware,” and “Download Factory Defaults” must be implemented in order, else the upgrade is incomplete.

Identifying Current Firmware Version

Before implementing a MultiVOIP firmware upgrade, be sure to verify the firmware version currently loaded on it. The firmware version appears in the MultiVoip Program menu. Go to **Start** | **Programs** | **MultiVOIP ____ x.xx**. The final expression, x.xx, is the firmware version number. In the illustration below, the firmware version is 4.00a, made for the E1 MultiVOIP (MVP3010).

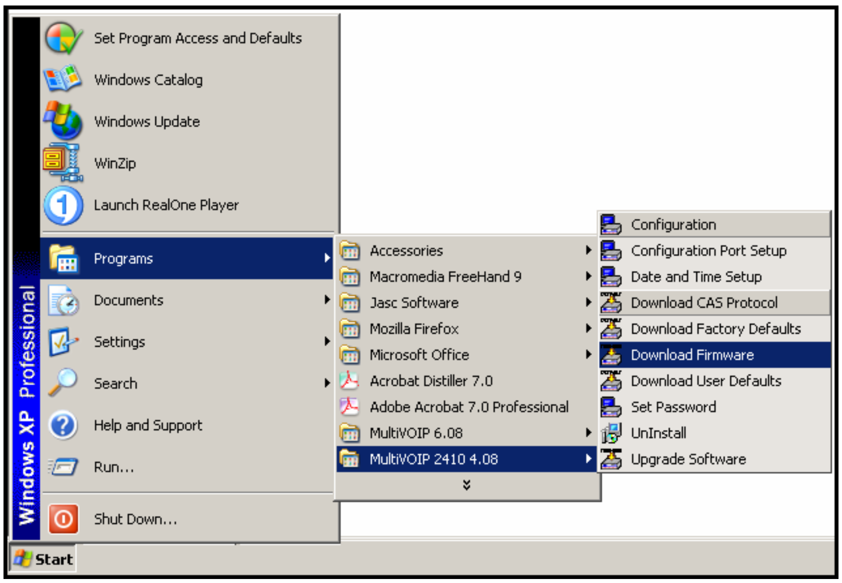
When a new firmware version is installed, the MultiVOIP software can be upgraded in one step using the **Upgrade Software** command, or piecemeal using the **Download Firmware** command and the **Download Factory Defaults** command.

Download Firmware transfers the firmware (including the H.323 protocol stack) in the PC's MultiVOIP directory into the nonvolatile flash memory of the MultiVOIP.
Download Factory Defaults sets all configuration parameters to the standard default values that are loaded at the MultiTech factory.
Upgrade Software implements both the **Download Firmware** command and the **Download Factory Defaults** command.

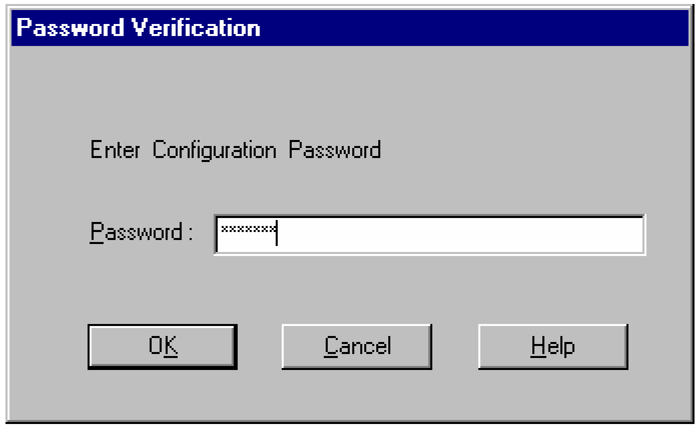
Downloading Firmware

1. The MultiVoip Configuration program must be off when invoking the **Download Firmware** command. If it is on, the command will not work.

2. To invoke the Download Factory Defaults command, go to **Start | Programs | MVP____ x.xx | Download Firmware**.

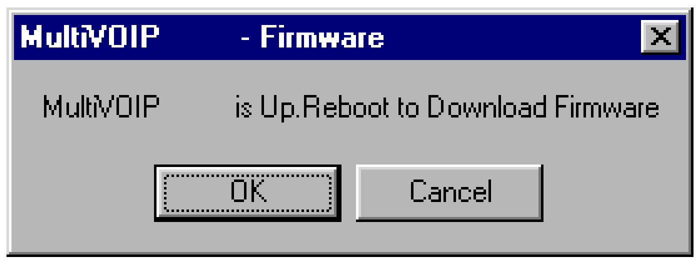


3. If a password has been established, the **Password Verification** screen will appear.



Type in the password and click **OK**.

4. The **MultiVOIP ___- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"

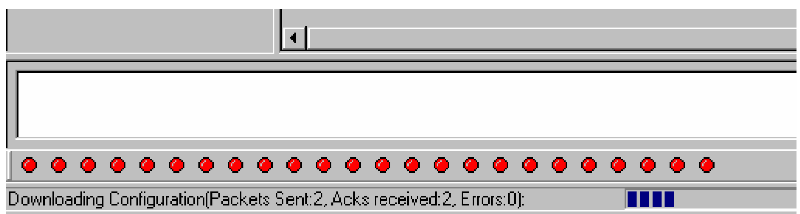


Click **OK** to download the firmware.

The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.

5. The program will locate the firmware ".bin" file in the MultiVOIP directory. Highlight the correct (newest) ".bin" file and click **Open**.

6. Progress bars will appear at the bottom of the screen during the file transfer.



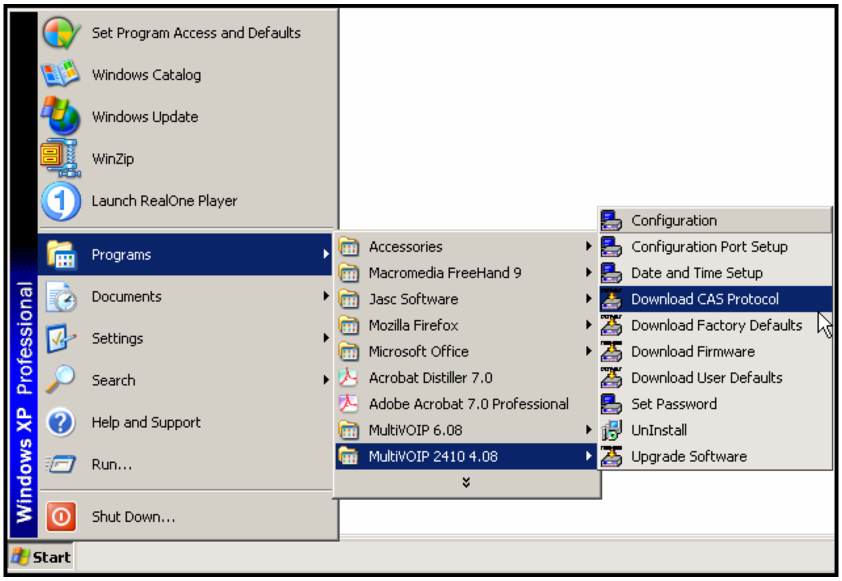
The MultiVOIP's "Boot" LED will turn off at the end of the transfer.

7. The **Download Firmware** procedure is complete.

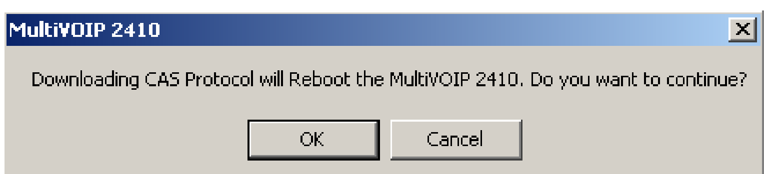
Downloading CAS Protocol

1. The MultiVoip Configuration program may be on or off when invoking the **Download CAS Protocol** command.

2.To invoke the **Download CAS Protocol** command, go to **Start | Programs | MVP____ x.xx | Download CAS Protocol**.

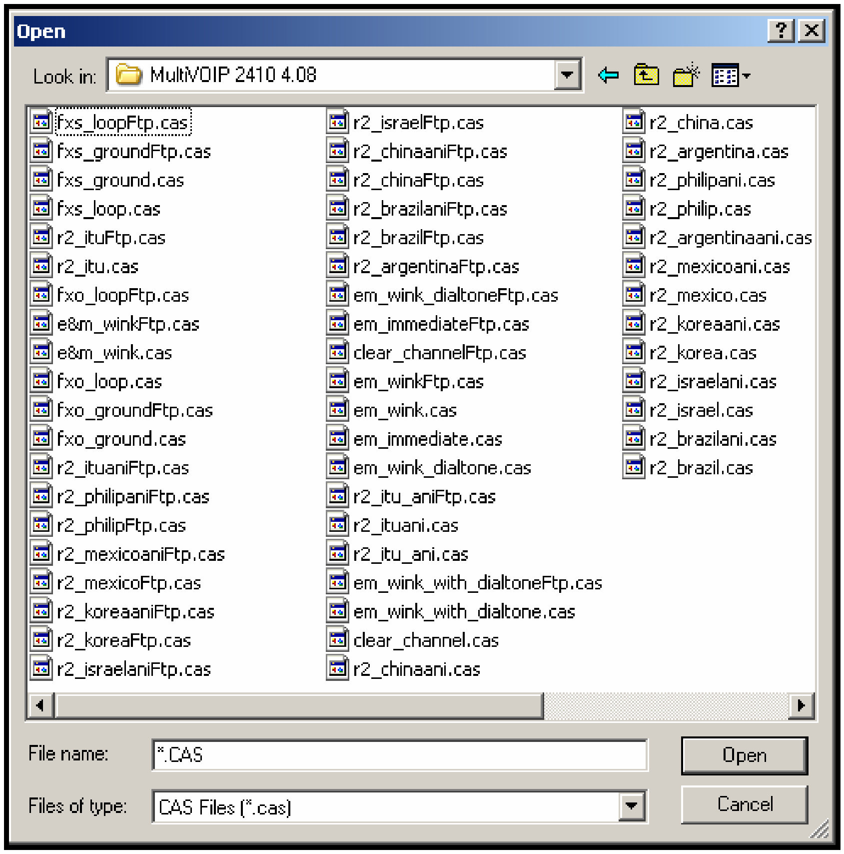


3. A message screen will appear warning that the download will entail a rebooting of the MultiVOIP.



Click **OK**.

4. The directory containing the CAS protocol files (extension is .cas) will appear.



Select the CAS protocol needed for your system. Click **Open**.

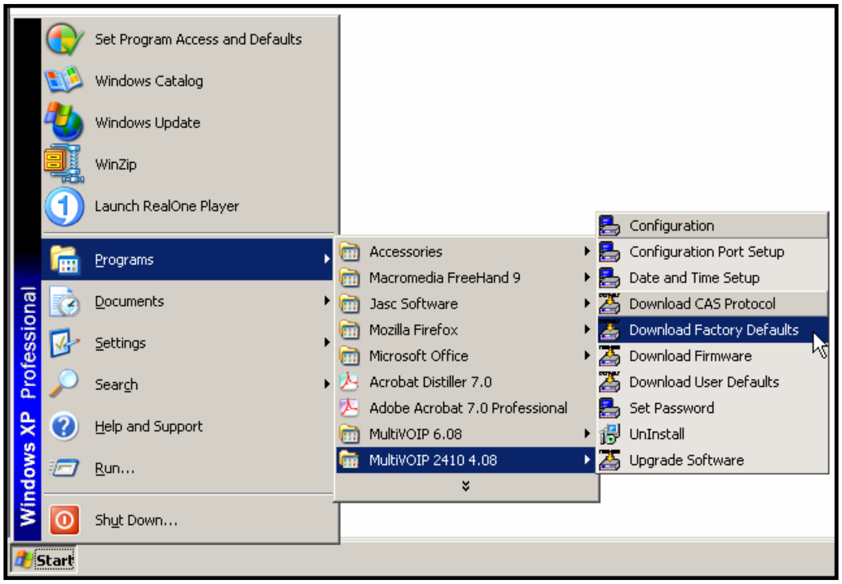
5. The chosen CAS protocol file will be loaded from the PC to the MultiVOIP unit. Progress bars will appear at the bottom of the screen while the download occurs. When the download is complete, the MultiVOIP will complete its rebooting process.

6. The MultiVOIP software will be closed when the download is complete. You will have to launch the MultiVOIP software again to continue using it.

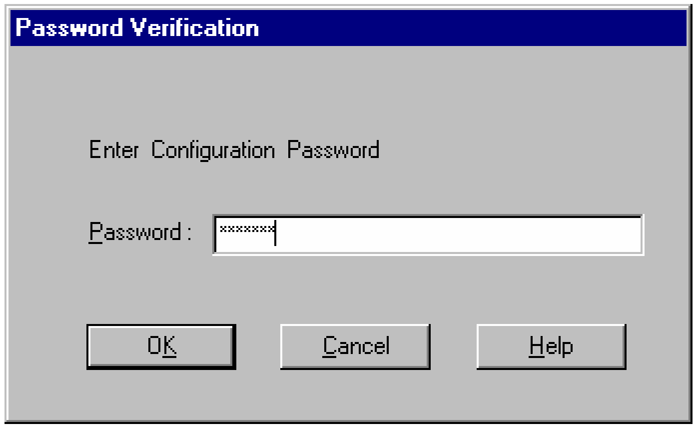
Downloading Factory Defaults

1. The MultiVoip Configuration program must be off when invoking the **Download Factory Defaults** command. If it is on, the command will not work.

2. To invoke the **Download Factory Defaults** command, go to **Start | Programs | MVP____ x.xx | Download Factory Defaults**.

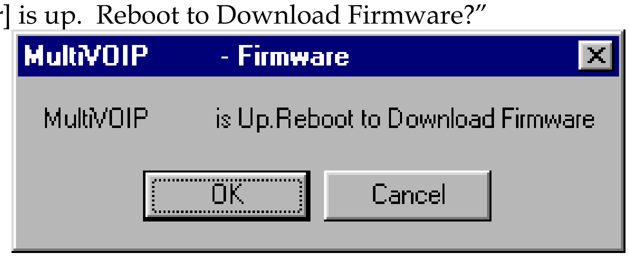


3. If a password has been established, the **Password Verification** screen will appear.



Type in the password and click **OK**.

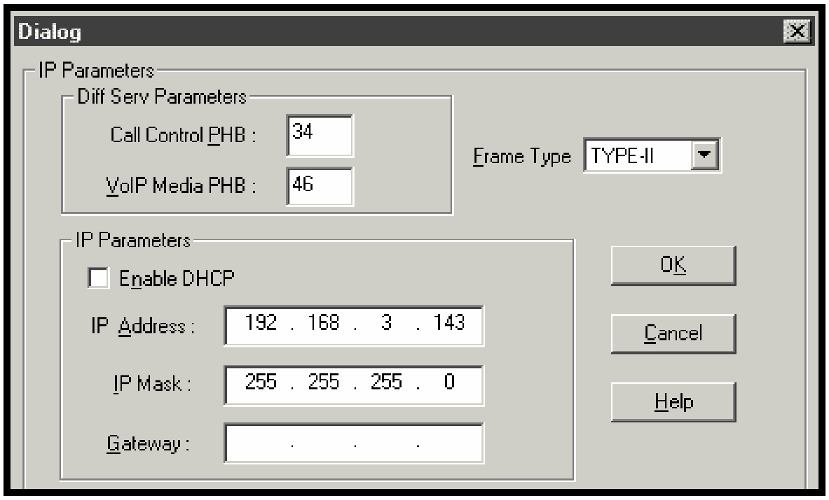
4. The **MVP____- Firmware** screen appears saying "MultiVOIP [*model number*] is up. Reboot to Download Firmware?"



Click **OK** to download the factory defaults.

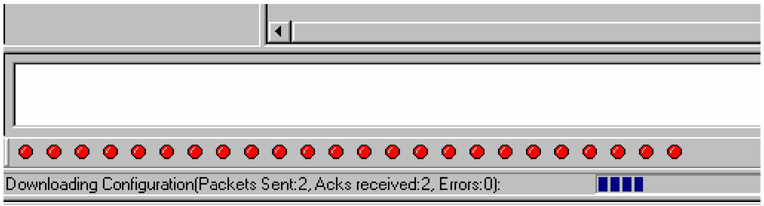
The "Boot" LED on the MultiVOIP will light up and remain lit during the file transfer process.

5. After the PC gets a response from the MultiVOIP, the **Dialog – IP Parameters** screen will appear.



The user should verify that the correct IP parameter values are listed on the screen and revise them if necessary. Then click **OK**.

6. Progress bars will appear at the bottom of the screen during the data transfer.



The MultiVOIP's "Boot" LED will turn off at the end of the transfer.

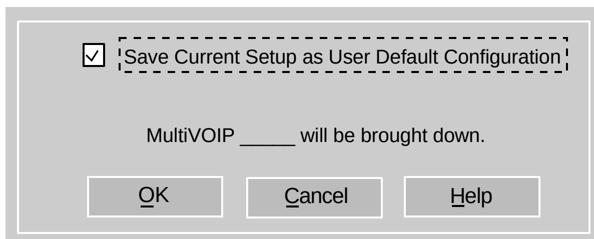
7. The **Download Factory Defaults** procedure is complete.

Setting and Downloading User Defaults

The **Download User Defaults** command allows you to maintain a known working configuration that is specific to your VOIP system. You can then experiment with alterations or improvements to the configurations confident that a working configuration can be restored if necessary.

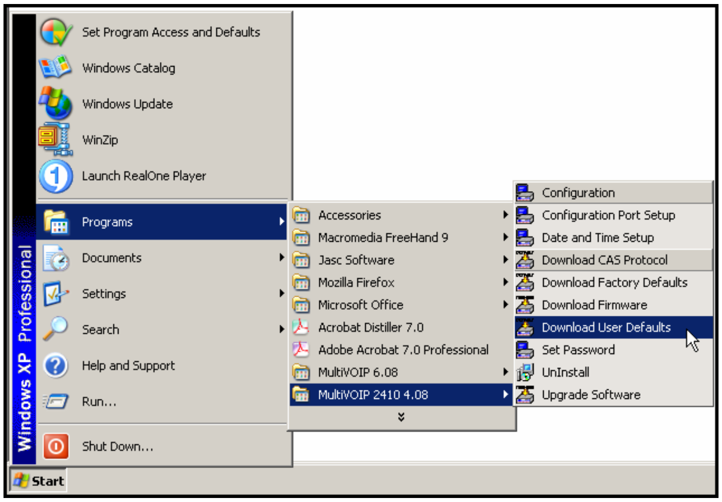
1. Before you can invoke the **Download User Defaults** command, you must first save a set of configuration parameters by using the **Save Setup** command in the sidebar menu of the MultiVOIP software.

2. Before the setup configuration is saved, you will be prompted to save the setup as the User Default Configuration. Select the checkbox and click **OK**.

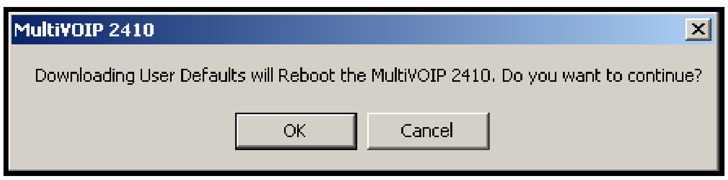


A user default file will be created. The MultiVOIP unit will reboot itself.

3. To download the user defaults, go to
 Start | Programs | MultiVOIP xxx | Download User Defaults.

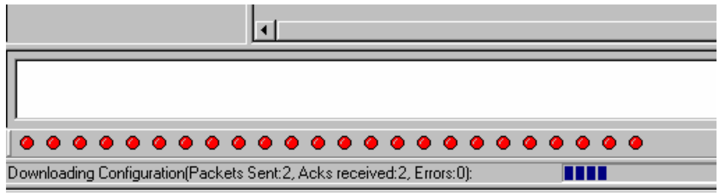


4. A confirmation screen will appear indicating that this action will
 entail rebooting the MultiVOIP.

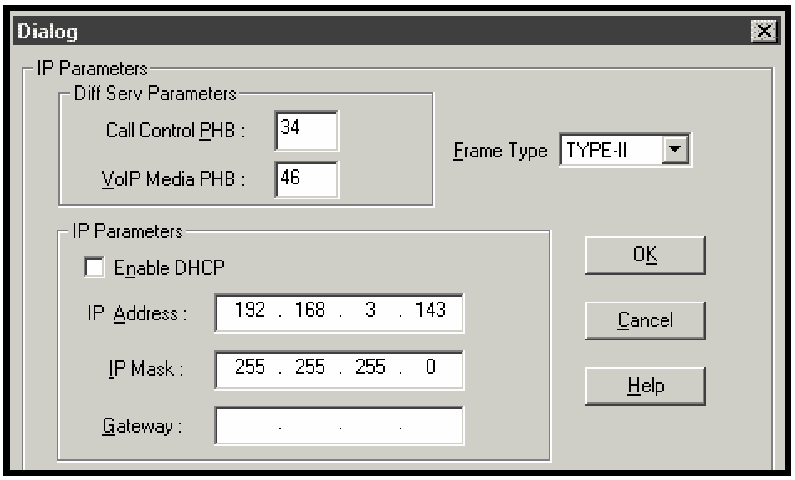


Click **OK**.

5. Progress bars will appear during the file transfer process.



5. When the file transfer process is complete, the **Dialog-- IP Parameters** screen will appear.



6. Set the IP values per your particular VOIP system. Click **OK**. Progress bars will appear as the MultiVOIP reboots itself.

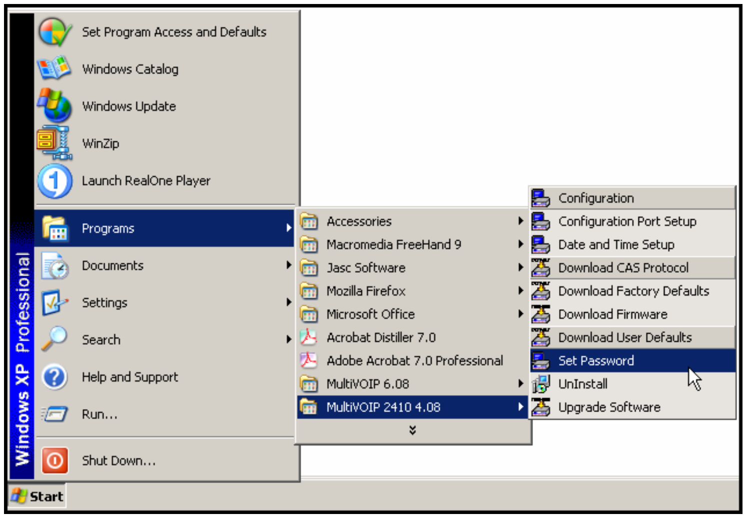
Setting a Password (Windows GUI)

After a user name has been designated and a password has been set, that password is required to gain access to any functionality of the MultiVOIP software. Only one user name and password can be assigned to a voip unit. The user name will be required when communicating with the MultiVOIP via the web browser GUI.

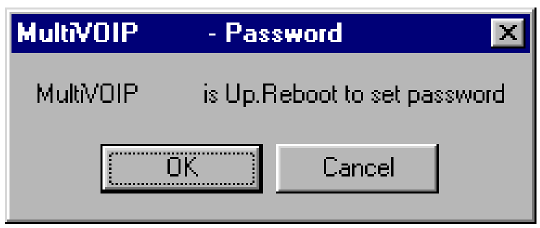
NOTE: Record your user name and password in a safe place. If the password is lost, forgotten, or unretrievable, the user must contact MultiTech Tech Support in order to resume use of the MultiVOIP unit.

1. The MultiVoip configuration program must be off when invoking the **Set Password** command. If it is on, the command will not work.

2. To invoke the **Set Password** command, go to **Start** | **Programs** | **MVP____ x.xx** | **Set Password**.



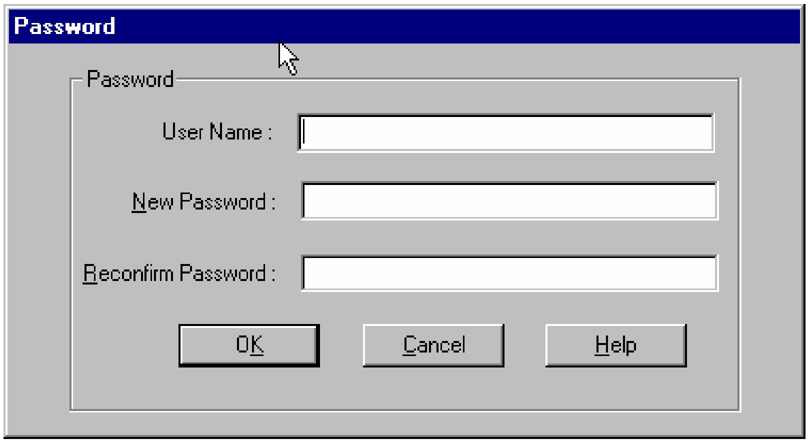
3. You will be prompted to confirm that you want to establish a password, which will entail rebooting the MultiVOIP (which is done automatically).



Click **OK** to proceed with establishing a password.

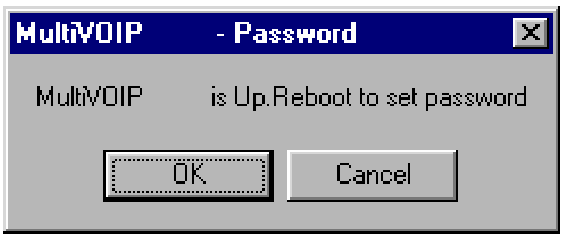
4. The **Password** screen will appear. If you intend to use the FTP Server function that is built into the MultiVOIP, enter a user name. (A User Name is not needed to access the local Windows GUI, the web browser GUI, or the commands in the **Program** group.) Type your password in the **Password** field of the **Password** screen. Type this same password again in the **Confirm Password** field to verify the password you have chosen.

NOTE: Be sure to write down your password in a convenient but secure place. If the password is forgotten, contact MultiTech Technical Support for advice.



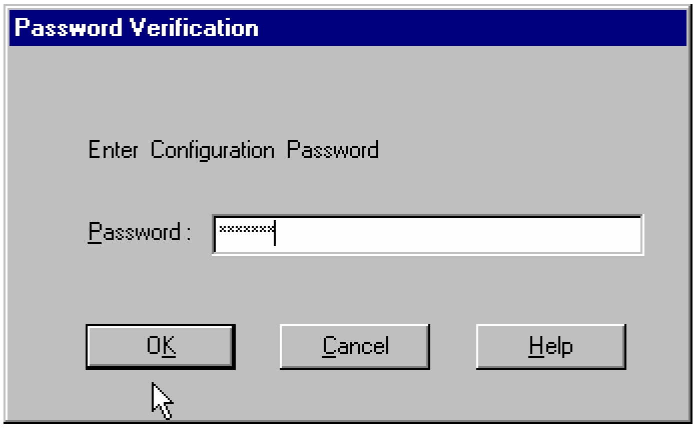
Click **OK**.

5. A message will appear indicating that a password has been set successfully.



After the password has been set successfully, the MultiVOIP will re-boot itself and, in so doing, its **BOOT** LED will light up.

6. After the password has been set, the user will be required to enter the password to gain access to the web browser GUI and any part of the MultiVOIP software listed in the **Program** group menu. User Name and Password are both needed for access to the FTP Server residing in the MultiVOIP.



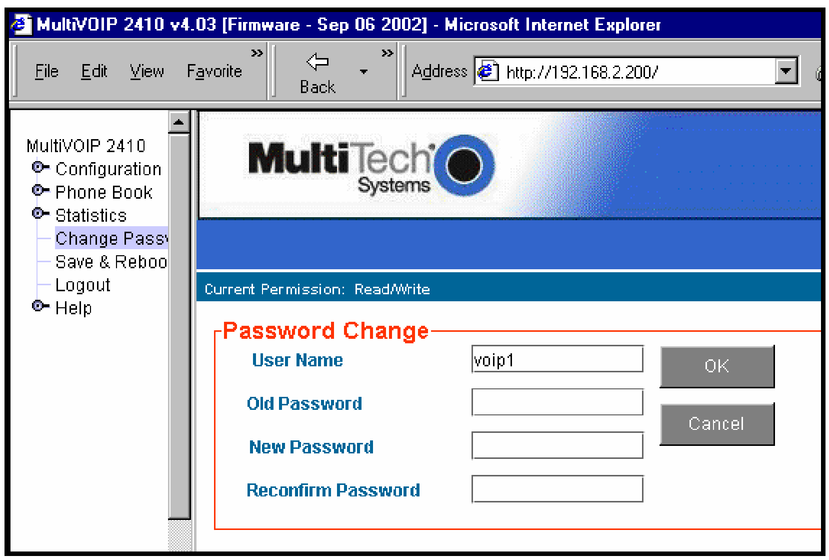
When MultiVOIP program asks for password at launch of program, the program will simply shut down if **CANCEL** is selected.

The MultiVOIP program will produce an error message if an invalid password is entered.

Setting a Password (Web Browser GUI)

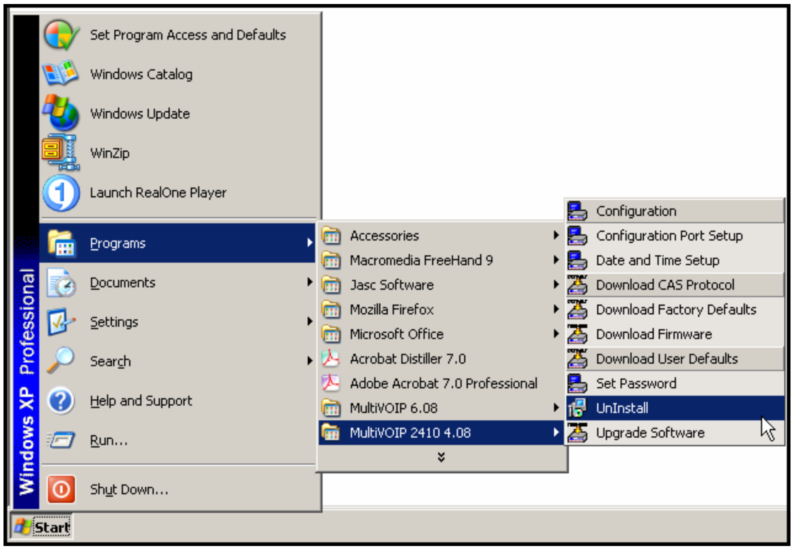
Setting a password is optional when using the MultiVOIP web browser GUI. Only one password can be assigned and it works for all MultiVOIP software functions (Windows GUI, web browser GUI, FTP server, and all Program menu commands, e.g., Upgrade Software – only the FTP Server function requires a User Name in addition to the password). After a password has been set, that password is required to access the MultiVOIP web browser GUI.

NOTE: Record your user name and password in a safe place. If the password is lost, forgotten, or unretrievable, the user must contact MultiTech Tech Support in order to resume use of the MultiVOIP web browser GUI.

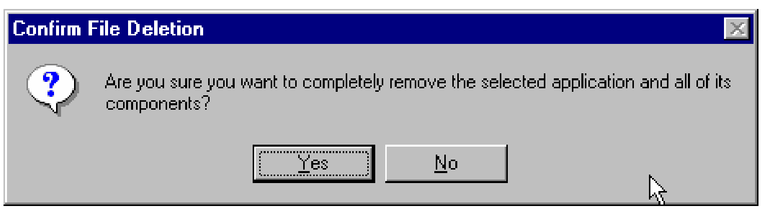


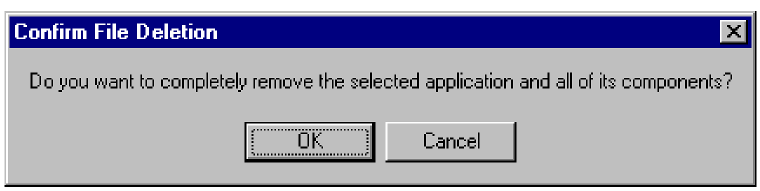
Un-Installing the MultiVOIP Software

1. To un-install the MultiVOIP configuration software, go to **Start** | **Programs** and locate the MultiVOIP entry. Select **Uninstall MVP____ vx.xx** (versions may vary).

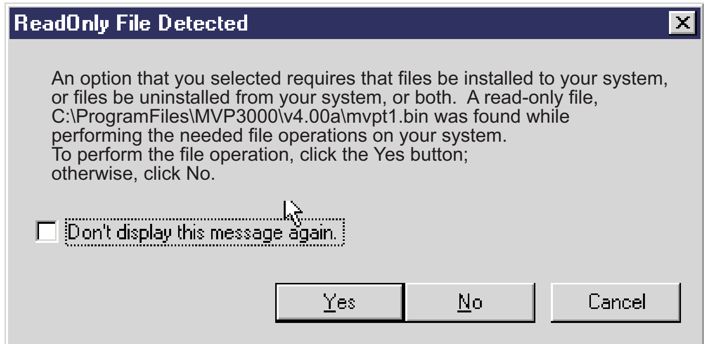


2. Two confirmation screens will appear. Click **Yes** and **OK** when you are certain you want to continue with the uninstallation process.

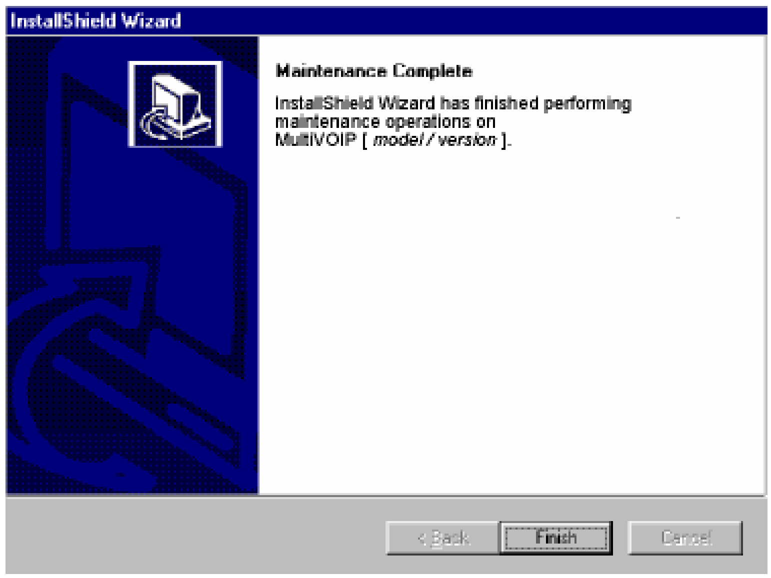




3. A special warning message similar to that shown below may appear for the MultiVOIP software's ".bin" file. Click **Yes**.



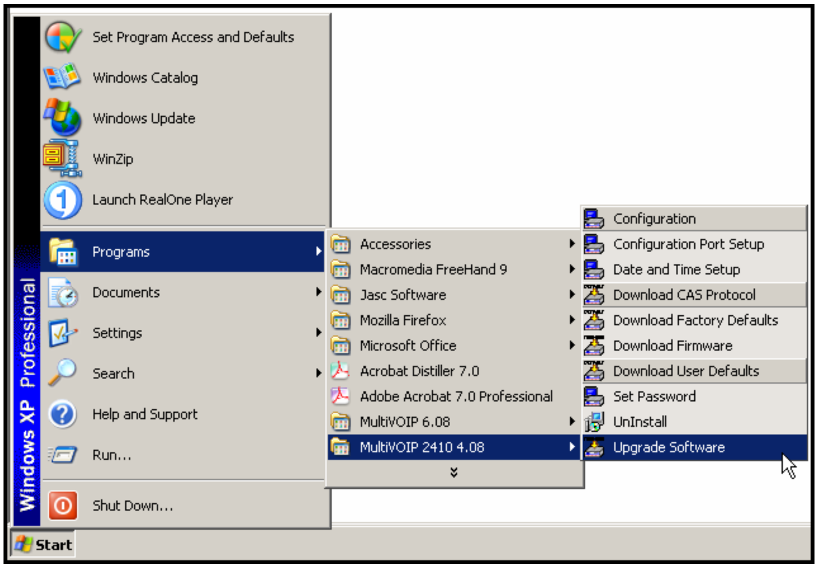
4. A completion screen will appear.



Click **Finish**.

Upgrading Software

As noted earlier (see the section *Implementing a Software Upgrade* above), the Upgrade Software command transfers, from the controller PC to the MultiVOIP unit, firmware (including the H.323 stack) and factory default configuration settings. As such, **Upgrade Software** implements the functions of both **Download Firmware** and **Download Factory Defaults** in a single command.



NOTE: To upgrade a MultiVOIP from software version 4.04 or earlier, an ftp primer file must first be sent to the VOIP. This file is located in the Software/ftp_Primer folder on the CD and the file name is "FTP_Primer.bin". Before uploading this file, it must be renamed "mvpt1ftp.bin". The VoIP will only accept files of this name. This is a safety precaution to prevent the wrong files from being uploaded to the VoIP. Once the primer file has been uploaded, upload the FTP firmware file. If you accepted the defaults during the software loading process, this file is located on your local drive at C:\Program Files\Multi-Tech Systems\MultiVOIP 4.08 where the X is the software number and the .08 is the version number of the MultiVOIP software on your local drive. Of course the firmware file is named 'mvpt1ftp.bin'.

Important: You cannot go back to 4.04 or earlier versions using FTP. You must use 'upgradesoftware' via the serial port.

Important: These ftp upgrade instructions do not apply to software release 4.05 and above.

FTP Server File Transfers ("Downloads")

MultiTech has built an FTP server into the MultiVOIP unit. Therefore, file transfers from the controller PC to the voip unit can be done using an FTP client program or even using a browser (e.g., Internet Explorer, Netscape or FireFox, used in conjunction with Windows Explorer).

The terminology of "downloads" and "uploads" gets a bit confusing in this context. File transfers from a client to a server are typically considered "uploads." File transfers from a large repository of data to machines with less data capacity are considered "downloads." In this case, these metaphors are contradictory: the FTP server is actually housed in the MultiVOIP unit, and the controller PC, which is actually the repository of the info to be transferred, uses an FTP client program. In this situation, we have chosen to call the transfer of files from the PC to the voip "downloads." (Be aware that some FTP client programs may use the opposite terminology, i.e., they may refer to the file transfer as an "upload ")

You can download firmware, CAS telephony protocols, default configuration parameters, and phonebook data for the MultiVOIP unit with this FTP functionality. These downloads are done over a network, not by a local serial port connection. Consequently, voips at distant locations can be updated from a central control point.

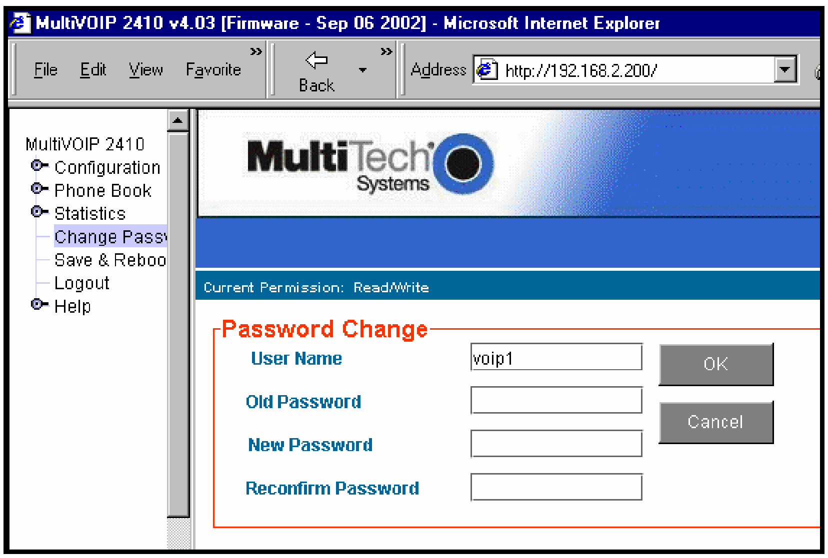
The phonebook downloading feature greatly reduces the data-entry required to establish inbound and outbound phonebooks for the voip units within a system. Although each MultiVOIP unit will require some unique phonebook entries, most will be common to the entire voip system. After the phonebooks for the first few voip units have been compiled, phonebooks for additional voips become much simpler: you copy the common material by downloading and then do data entry for the few phonebook items that are unique to that particular voip unit or voip site.

To transfer files using the FTP server functionality in the MultiVOIP, follow these directions.

1. **Establish Network Connection and IP Addresses**. Both the controller PC and the MultiVOIP unit(s) must be connected to the same IP network. An IP address must be assigned for each.

IP Address of Control PC	____ .	____ .	____ .	____
IP Address of voip unit #1	____ .	____ .	____ .	____
:	:	:	:	:
IP address of voip unit #n	____ .	____ .	____ .	____

2. **Establish User Name and Password**. You must establish a user name and (optionally) a password for contacting the voip over the IP network. (When connection is made via a local serial connection between the PC and the voip unit, no user name is needed.)

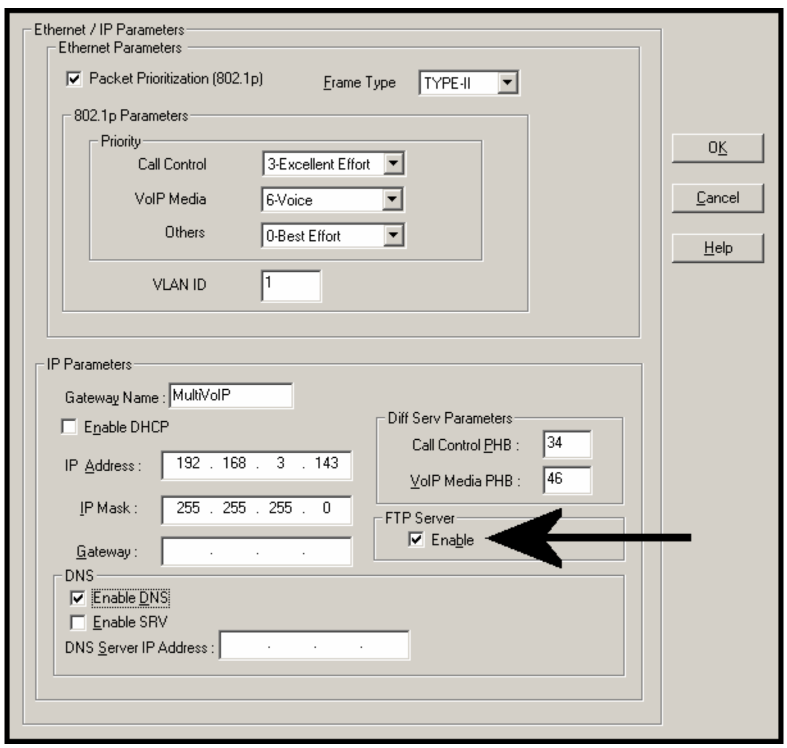


As shown above, the username and password can be set in the web GUI as well as in the Windows GUI.

3. **Install FTP Client Program or Use Substitute**. You *should* install an FTP client program on the controller PC. FTP file transfers can be done using a web browser (e.g., Netscape or Internet Explorer) in conjunction with a local Windows browser a (e.g., Windows Explorer), but this approach is somewhat clumsy (it requires use of two application programs rather than one) and it limits downloading to only one VOIP unit at a time. With an FTP client program, multiple voips can receive FTP file transmissions in response to a single command (the transfers may occur serially however).

Although MultiTech does not provide an FTP client program with the MultiVOIP software or endorse any particular FTP client program, we remind our readers that adequate FTP programs are readily available under retail, shareware and freeware licenses. (Read and observe any End-User License Agreement carefully.) Two examples of this are the "WSFTP" client and the "SmartFTP" client, with the former having an essentially text-based interface and the latter having a more graphically oriented interface, as of this writing. User preferences will vary. Examples here show use of both programs.

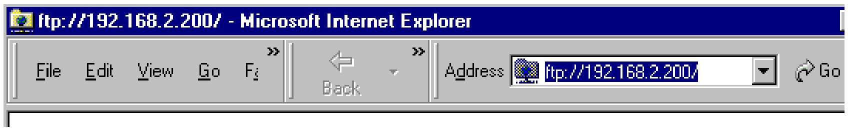
4. **Enable FTP Functionality**. Go to the **Ethernet/IP Parameters** screen and click on the "FTP Server: Enable" box.



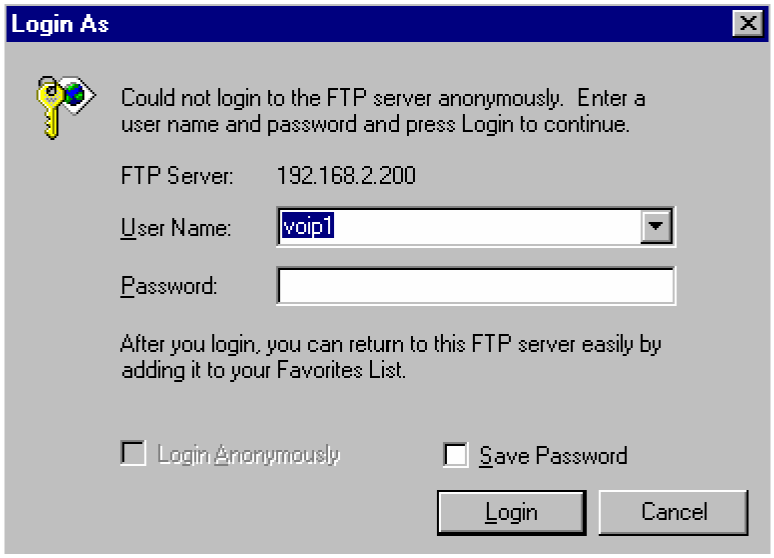
5. **Identify Files to be Updated**. Determine which files you want to update. Six types of files can be updated using the FTP feature. In some cases, the file to be transferred will have "Ftp" as the part of its filename just before the suffix (or extension). So, for example, the file "mvpt1Ftp.bin" can be transferred to update the bin file (firmware) residing in the MultiVOIP. Similarly, the file "fxo_loopFtp.cas" could be transferred to enable use of the FXO Loop Start telephony interface in one of the analog voip units and the file "r2_brazilFtp.cas" could be transferred to enable a particular telephony protocol used in Brazil.

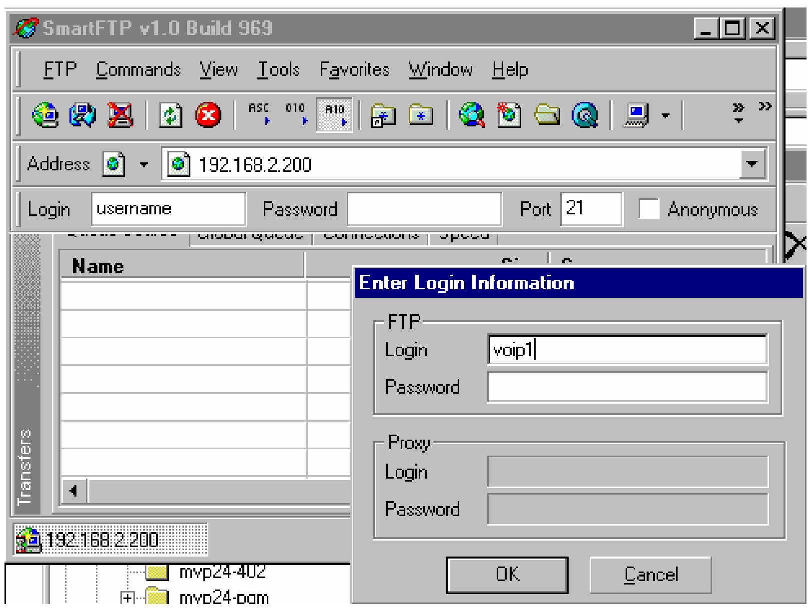
File Type	File Names	Description
firmware "bin" file	mvpt1Ftp.bin	This is the MultiVOIP firmware file. Only one file of this type will be in the directory.
factory defaults	fdefFtp.cnf	This file contains factory default settings for user-changeable configuration parameters. Only one file of this type will be in the directory.
CAS file	fxo_loopFtp.cas, em_winkFtp.cas, r2_brazilFtp.cas r2_chinaFtp.cas	These telephony files are for Channel Associated Signaling. The directory contains many CAS files, some labeled for specific functionality, others for countries or regions where certain attributes are standard. Any CAS file used must first be renamed to "CASFILE.CAS."
inbound phonebook	InPhBk.tmr	This file updates the inbound phonebook in the MultiVOIP unit.
outbound phonebook	OutPhBk.tmr	This file updates the outbound phonebook in the MultiVOIP unit.

6. **Contact MultiVOIP FTP Server**. You must make contact with the FTP Server in the voip using either a web browser or FTP client program. Enter the IP address of the MultiVOIP's FTP Server. If you are using a browser, the address must be preceded by "ftp://" (otherwise you'll reach the web GUI within the MultiVOIP unit).



7. **Log In**. Use the User Name and password established in item #2 above. The login screens will differ depending on whether the FTP file transfer is to be done with a web browser (see first screen below) or with an FTP client program (see second screen below).



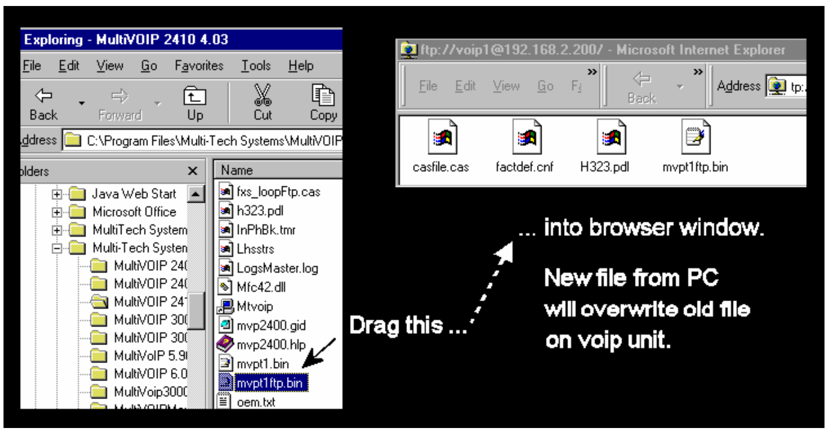


8. **Invoke Download**. Downloading can be done with a web browser or with an FTP client program.

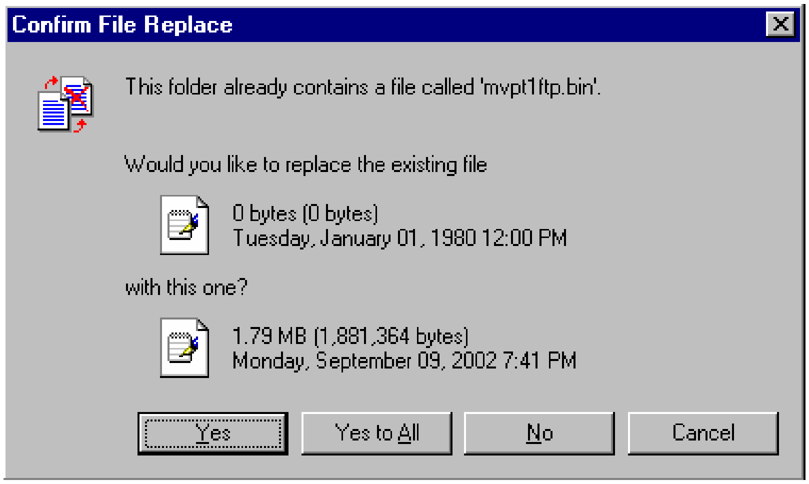
8A. Download with Web Browser.

8A1. In the local Windows browser, locate the directory holding the MultiVOIP program files. The default location will be C:\Program Files \Multi-Tech Systems \MultiVOIP xxxx yyyy (where x and y represent MultiVOIP model numbers and software version numbers).

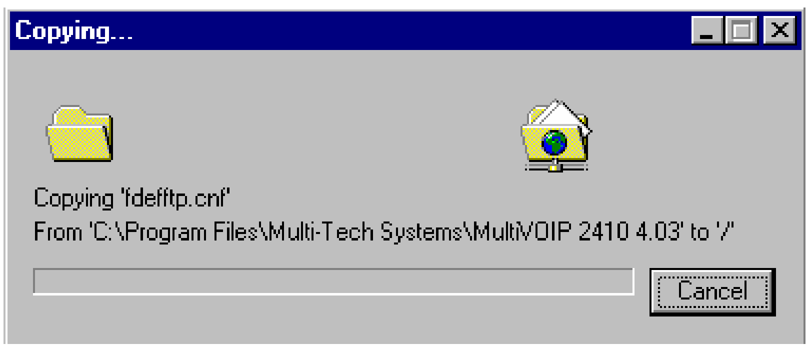
8A2. Drag-and-drop files from the local Windows browser (e.g., Windows Explorer) to the web browser.



You may be asked to confirm the overwriting of files on the MultiVOIP. Do so.



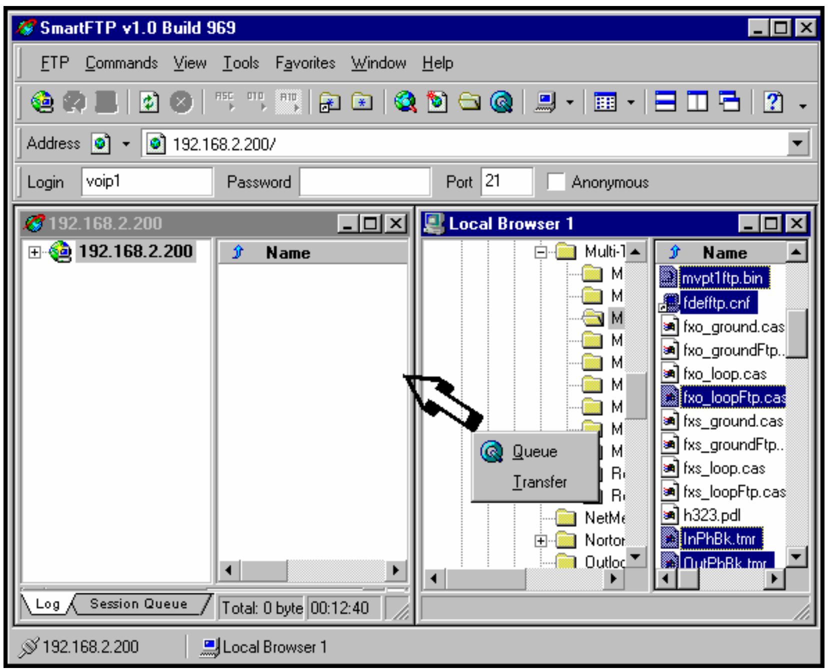
File transfer between PC and voip will look like transfer within voip directories.



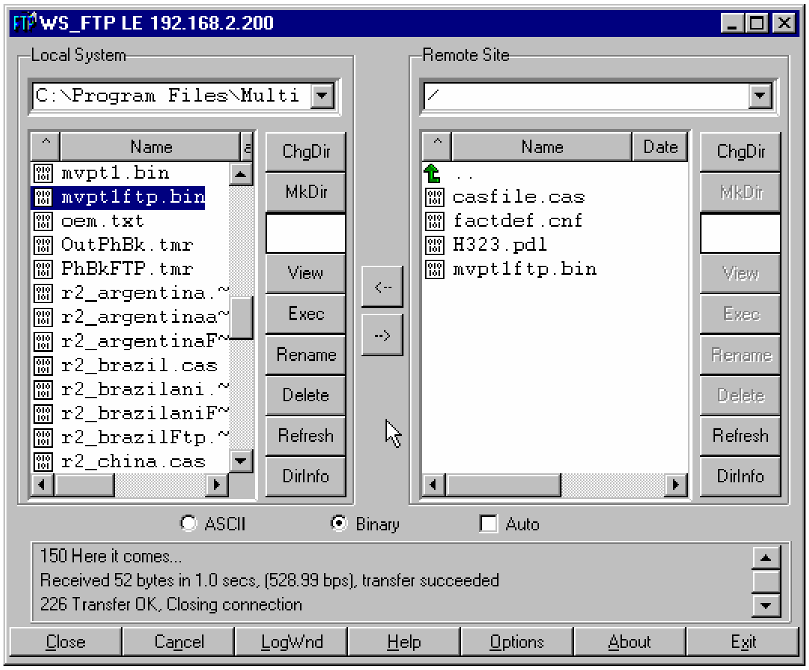
8B. Download with FTP Client Program.

8B1. In the local directory browser of the FTP client program, locate the directory holding the MultiVOIP program files. The default location will be C:\Program Files \Multi-Tech Systems \MultiVOIP xxxx yyyy (where x and y represent MultiVOIP model numbers and software version numbers).

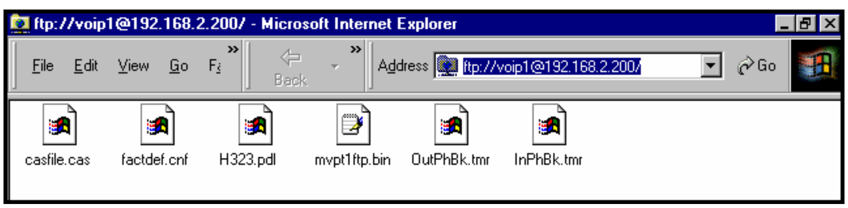
8B2. In the FTP client program window, drag-and-drop files from the local browser pane to the pane for the MultiVOIP FTP server. FTP client GUI operations vary. In some cases, you can choose between immediate and queued transfer. In some cases, there may be automated capabilities to transfer to multiple destinations with a single command.



Some FTP client programs are more graphically oriented (see previous screen), while others (like the "WS-FTP" client) are more text oriented.

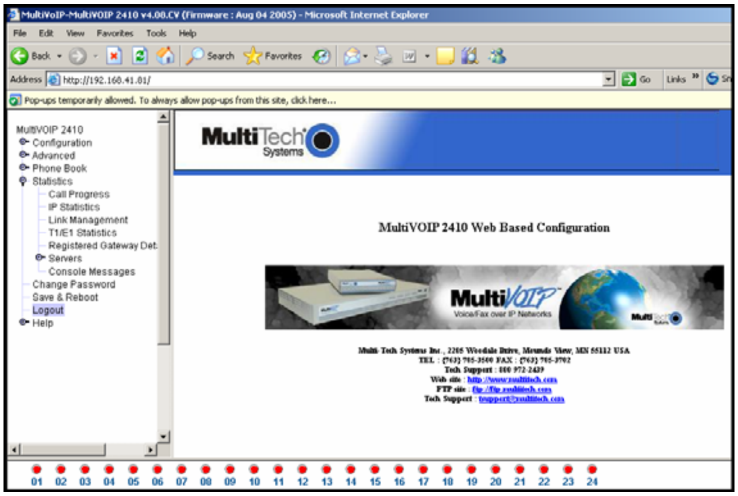


9. **Verify Transfer**. The files transferred will appear in the directory of the MultiVOIP.



10. **Log Out of FTP Session**. Whether the file transfer was done with a web browser or with an FTP client program, you *must* log out of the FTP session before opening the MultiVOIP Windows GUI.

Web Browser Interface



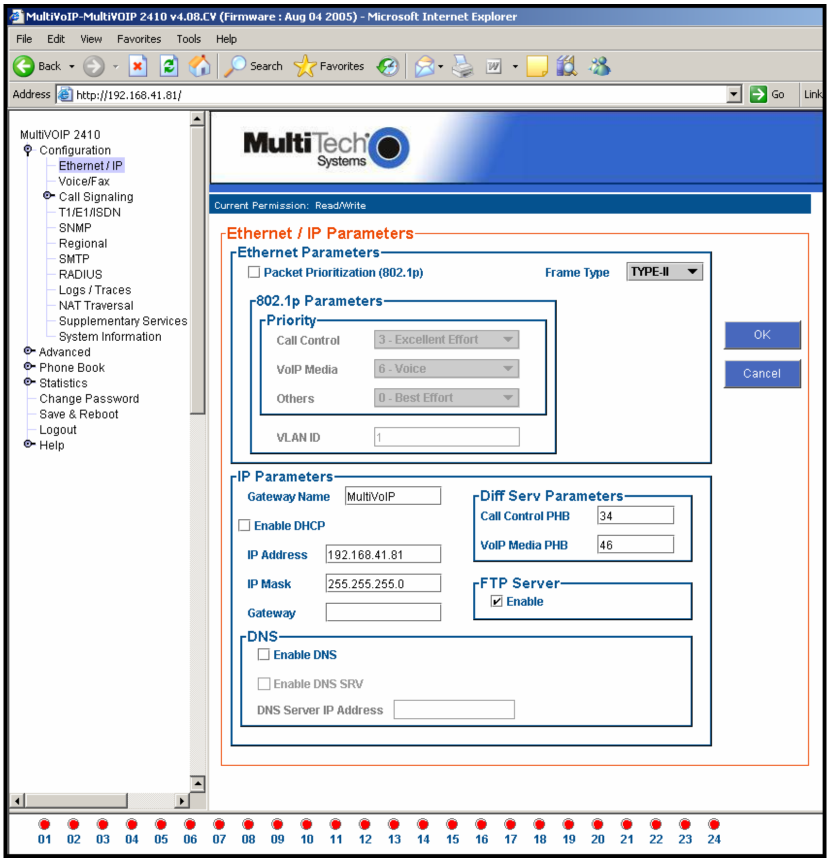
You can control the MultiVOIP unit with a graphic user interface (GUI) based on the common web browser platform. Qualifying browsers are InternetExplorer6, Netscape6, and Mozilla FireFox 1.0.

Pop-Ups. Note that the MultiVOIP Web GUI uses pop-up windows extensively. You must configure the browser to allow pop-ups when using the MultiVOIP Web GUI.

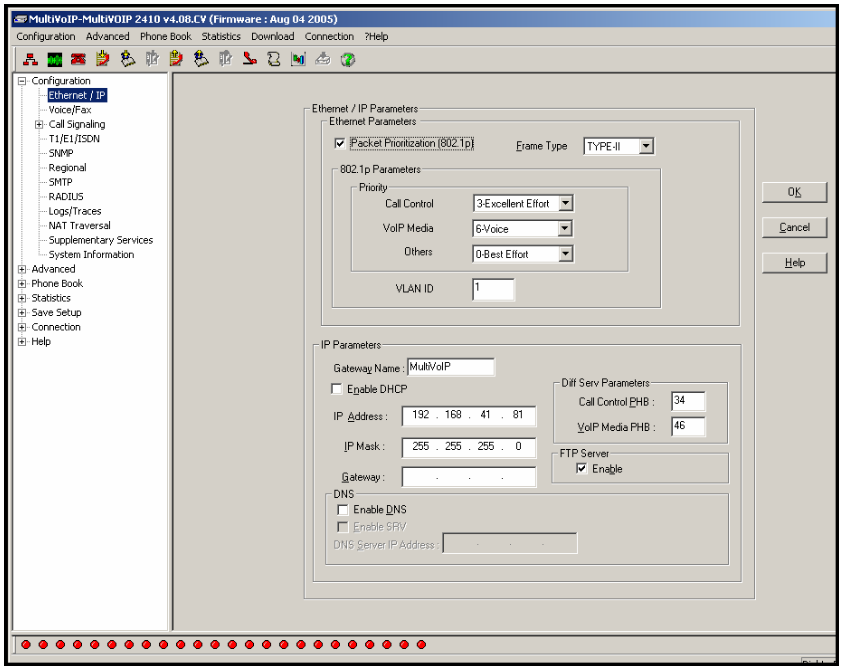
MultiVOIP Web Browser GUI Overview	
Function	Remote configuration and control of MultiVOIP units.
Configuration Prerequisite	Local Windows GUI must be used to assign IP address to MultiVOIP.
Browser Version Requirement	Internet Explorer 6.0 or higher; or Netscape 6.0 or higher; or Mozilla Firefox 1.0 or higher
Java Requirement	Java Runtime Environment version 1.4.0_01 or higher (this application program is included with MultiVOIP)
Video Usability	large video monitor recommended

The initial configuration step of assigning the voip unit an IP address must still be done locally using the Windows GUI. However, all additional configuration can be done via the web GUI.

The content and organization of the web GUI is directly parallel to the Windows GUI. For each screen in the Windows GUI, there is a corresponding screen in the web GUI. The fields on each screen are the same, as well.

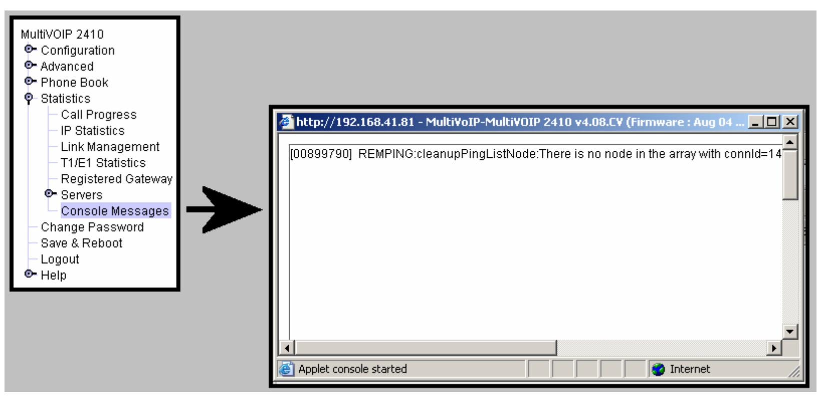


The Windows GUI gives access to commands via icons and pulldown menus whereas the web GUI does not.



The web GUI cannot perform logging in the same direct mode done in the Windows GUI. However, when the web GUI is used, logging can be done by email (SMTP).

The web GUI gives easy access to **Console Messages**. Whereas with the Windows GUI console messages must be viewed using a communications program like HyperTerminal, with the Web GUI, it's easy: just click on **STATISTICS | CONSOLE MESSAGES** and a pop-up window appears.

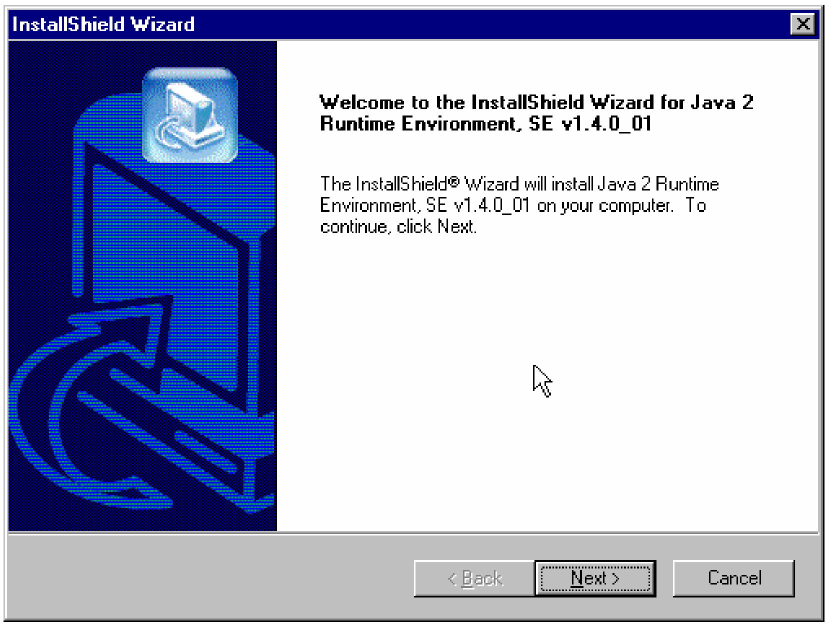


The graphic layout of the web GUI is also somewhat larger-scale than that of the Windows GUI. For that reason, it's helpful to use as large of a video monitor as possible in order to see all of a screen's contents with minimal scrolling.

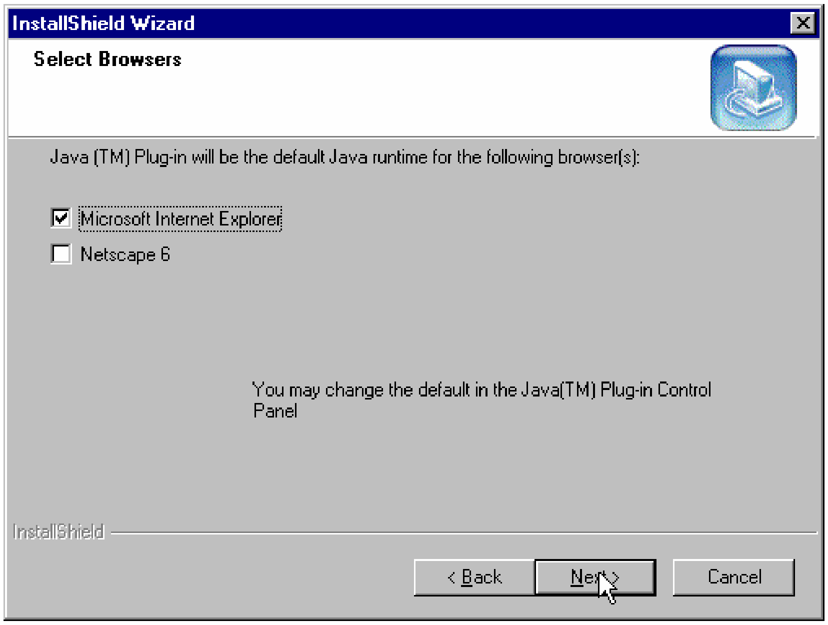
The primary advantage of the web GUI is remote access for control and configuration. The controller PC and the MultiVOIP unit itself must both be connected to the same IP network and their IP addresses must be known.

In order to use the web GUI, you must also install a Java application program on the controller PC. This Java program is included on the MultiVOIP product CD.). Java is needed to support drop-down menus and multiple windows in the web GUI.

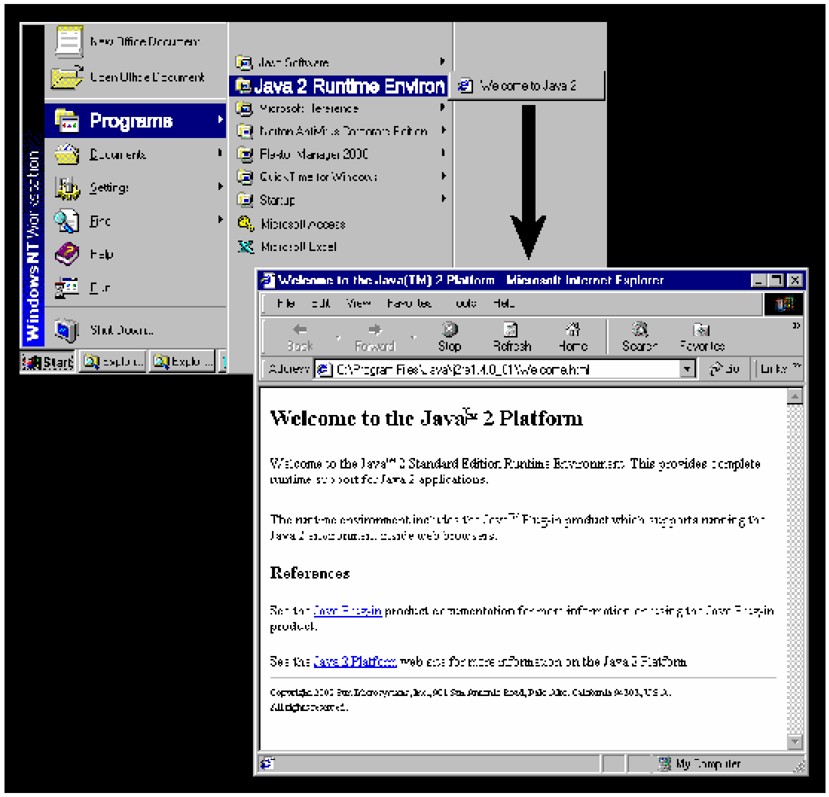
To install the Java program, go to the **Java** directory on the MultiVOIP product CD. Double-click on the EXE file to begin the installation. Follow the instructions on the Install Shield screens.



During the installation, you must specify which browser you'll use in the **Select Browsers** screen.



When installation is complete, the Java program becomes accessible in your **Start | Programs** menu (Java resources are readily available via the web). However, the Java program runs automatically in the background as a plug-in supporting the MultiVOIP web GUI. No overt user actions are required.



After the Java program has been installed, you can access the MultiVOIP using the web browser GUI. Close the MultiVOIP Windows GUI. Start the web browser. Enter the IP address of the MultiVOIP unit. Enter a password when prompted. (A password is needed here only if password has been set for the local Windows GUI or for the MultiVOIP's FTP Server function. See "Setting a Password -- Web Browser GUI" earlier in this chapter.) The web browser GUI offers essentially the same control over the voip as can be achieved using the Windows GUI. As noted earlier, logging functions cannot be handled via the web GUI. And, because network communications will be slower than direct communications over a serial PC cable, command execution will be somewhat slower over the web browser GUI than with the Windows GUI.

SysLog Server Functions

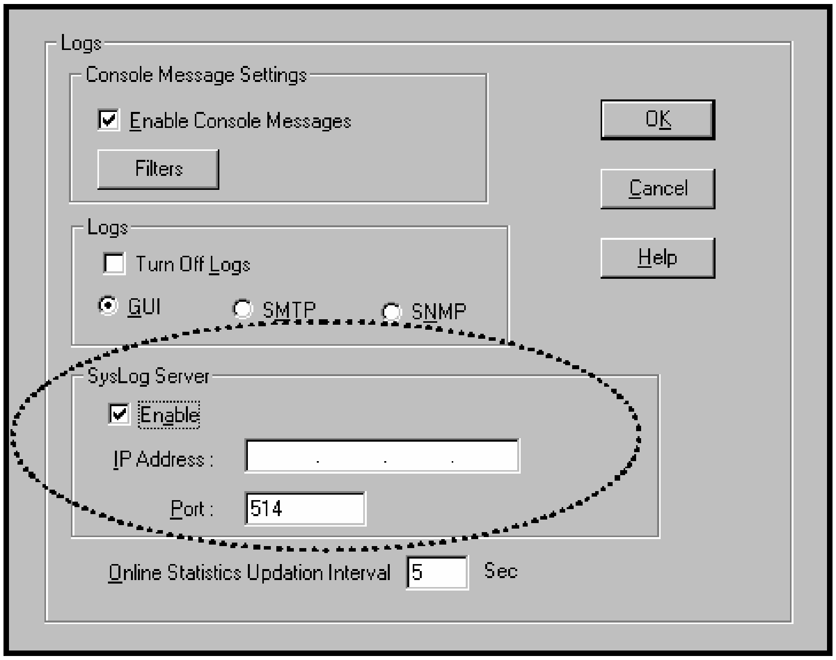
MultiTech has built SysLog server functionality into the software of the MultiVOIP units. SysLog is a *de facto* standard for logging events in network communication systems.

The SysLog Server resides in the MultiVOIP unit itself. To implement this functionality, you will need a SysLog client program (sometimes referred to as a "daemon"). SysLog client programs, both paid and freeware, can be obtained from Kiwi Enterprises, among other firms. Read the End-User License Agreement carefully and observe license requirements. See www.kiwisyslog.com. SysLog client programs essentially give you a means of structuring console messages for convenience and ease of use.

MultiTech Systems does not endorse any particular SysLog client program. SysLog client programs by qualified providers should suffice for use with MultiVOIP units. Kiwi's brief description of their SysLog program is as follows:

> "Kiwi Syslog Daemon is a freeware Syslog Daemon for the Windows platform. It receives, logs, displays and forwards Syslog messages from hosts such as routers, switches, Unix hosts and any other syslog enabled device. There are many customizable options available."

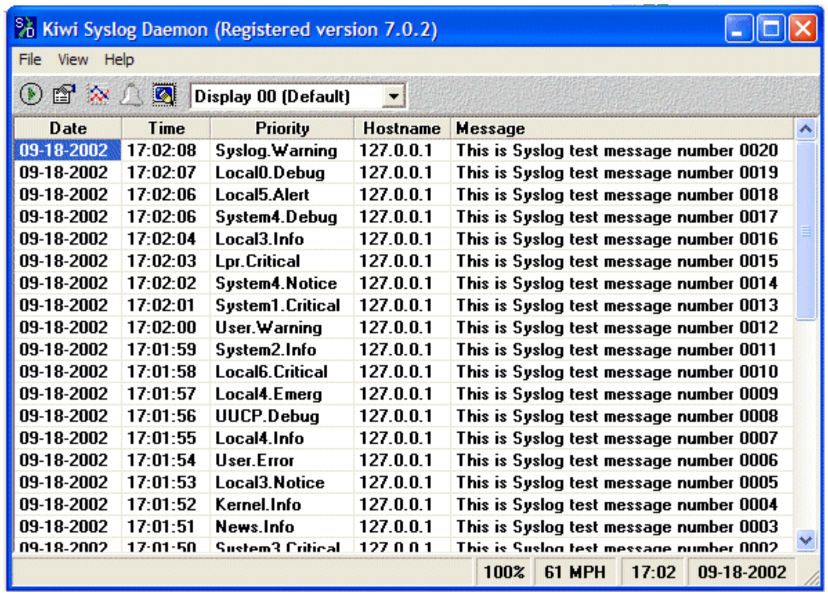
Before a SysLog client program is used, the SysLog functionality must be enabled within the MultiVOIP in the **Logs** menu under **Configuration**.



The IP Address used will be that of the MultiVOIP itself.

In the **Port** field, entered by default, is the standard ('well-known') logical port, 514.

Configuring the SysLog Client Program. Configure the SysLog client program for your own needs. In various SysLog client programs, you can define where log messages will be saved/archived, opt for interaction with an SNMP system (like MultiVoipManager), set the content and format of log messages, determine disk space allocation limits for log messages, and establish a hierarchy for the seriousness of messages (normal, alert, critical, emergency, etc.). A sample presentation of SysLog info in the Kiwi daemon is shown below. SysLog programs will vary in features and presentation.



Chapter 9 Warranty, Service, and Tech Support

Limited Warranty

Multi-Tech Systems, Inc. ("MTS") warrants that its products will be free from defects in material or workmanship for a period of two years from the date of purchase, or if proof of purchase is not provided, two years from date of shipment. MTS MAKES NO OTHER WARRANTY, EXPRESSED OR IMPLIED, AND ALL IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE HEREBY DISCLAIMED. This warranty does not apply to any products which have been damaged by lightning storms, water, or power surges or which have been neglected, altered, abused, used for a purpose other than the one for which they were manufactured, repaired by the customer or any party without MTS's written authorization, or used in any manner inconsistent with MTS's instructions.

MTS's entire obligation under this warranty shall be limited (at MTS's option) to repair or replacement of any products which prove to be defective within the warranty period, or, at MTS's option, issuance of a refund of the purchase price. Defective products must be returned by Customer to MTS's factory—transportation prepaid.

MTS WILL NOT BE LIABLE FOR CONSEQUENTIAL DAMAGES AND UNDER NO CIRCUMSTANCES WILL ITS LIABILITY EXCEED THE PURCHASE PRICE FOR DEFECTIVE PRODUCTS.

Repair Procedures for U.S. and Canadian Customers

In the event that service is required, products may be shipped, freight prepaid, to our Mounds View, Minnesota factory:

Multi-Tech Systems, Inc.
2205 Woodale Drive
Mounds View, MN 55112
Attn: Repairs, Serial # ________________

A Returned Materials Authorization (RMA) is not required. Return shipping charges (surface) will be paid by MTS.

Please include, inside the shipping box, a description of the problem, a return shipping address (it must be a street address, not a P.O. Box number), your telephone number, and if the product is out of warranty, a check or purchase order for repair charges.

For out-of-warranty repair charges, go to www.multitech.com/documents/warranties

Extended two-year overnight replacement service agreements are available for selected products. Please call MTS at (888) 288-5470, extension 5308, or visit our web site at www.multitech.com/programs/orc for details on rates and coverages.

Please direct your questions regarding technical matters, product configuration, verification that the product is defective, etc., to our Technical Support department at (800) 972-2439 or email tsupport@multitech.com. Please direct your questions regarding repair expediting, receiving, shipping, billing, etc., to our Repair Accounting department at (800) 328-9717 or (763) 717-5631, or email mtsrepair@multitech.com.

Repairs for damages caused by lightning storms, water, power surges, incorrect installation, physical abuse, or used-caused damages are billed on a time-plus-materials basis.

Technical Support

Multi-Tech Systems has an excellent staff of technical support personnel available to help you get the most out of your Multi-Tech product. If you have any questions about the operation of this unit, or experience difficulty during installation you can contact Tech Support via the following:

Contacting Technical Support

Country	By E-mail	By telephone
France	support@multitech.fr	(33) 1-64 61 09 81
India	support@ multitechindia.com	(91) 124-340778
U.K.	support@ multitech.co.uk	(44) 118 959 7774
U.S. & Canada	tsupport@ multitech.com	(800) 972-2439
Rest of World	support@ multitech.com	(763) 785-3500

Internet: http://www.multitech.com/ **_forms/email_tech_support.htm**

Please have your product information available, including model and serial number.

Chapter 10: Regulatory Information

EMC, Safety, and R&TTE Directive Compliance

The CE mark is affixed to this product to confirm compliance with the following European Community Directives:

Council Directive 89/336/EEC of 3 May 1989 on the approximation of the laws of Member States relating to electromagnetic compatibility,
and
Council Directive 73/23/EEC of 19 February 1973 on the harmonization of the laws of Member States relating to electrical equipment designed for use within certain voltage limits,
and
Council Directive 1999/5/EC of 9 March 1999 on radio equipment and telecommunications terminal equipment and the mutual recognition of their conformity.

FCC Declaration

NOTE: This equipment has been tested and found to comply with the limits for a **Class A** digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy, and if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

This device complies with Part 15 of the FCC rules.

Operation is subject to the following two conditions:

(1) This device may not cause harmful interference.

(2) This device must accept any interference that may cause undesired operation.

Warning: Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

Industry Canada

This Class A digital apparatus meets all requirements of the Canadian Interference-Causing Equipment Regulations.

Cet appareil numérique de la classe A
respecte toutes les exigences du
Reglement Canadien sur le matériel brouilleur.

FCC Part 68 Telecom

1. This equipment complies with part 68 of the Federal Communications Commission Rules. On the outside surface of this equipment is a label that contains, among other information, the FCC registration number. This information must be provided to the telephone company.
2. As indicated below, the suitable jack (Universal Service Order Code connecting arrangement) for this equipment is shown. If applicable, the facility interface codes (FIC) and service order codes (SOC) are shown.
3. An FCC compliant telephone cord and modular plug is provided with this equipment. This equipment is designed to be connected to the telephone network or premises wiring using a compatible modular jack that is Part 68 compliant. See installation instructions for details.
4. If this equipment causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. If advance notice is not practical, the telephone company will notify the customer as soon as possible.
5. The telephone company may make changes in its facilities, equipment, operation, or procedures that could affect the operation of the equipment. If this happens, the telephone company will provide advance notice to allow you to make necessary modifications to maintain uninterrupted service.
6. If trouble is experienced with this equipment (the model of which is indicated below), please contact Multi-Tech Systems, Inc. at the address shown below for details of how to have repairs made. If the equipment is causing harm to the network, the telephone company

may request you to remove the equipment form t network until the problem is resolved.

7. No repairs are to be made by you. Repairs are to be made only by Multi-Tech Systems or its licensees. Unauthorized repairs void registration and warranty.

8. | | |
|---|---|
| Manufacturer: | Multi-Tech Systems, Inc. |
| Trade name: | MultiVOIP |
| Model number: | MVP-810/410/210 |
| FCC registration number: | US: AU7DDNAN46050 |
| Modular jack (USOC): | RJ-48C |
| Service center in USA: | Multi-Tech Systems, Inc.
2205 Woodale Drive
Mounds View, MN 55112
Tel: (763) 785-3500
FAX: (763) 785-9874 |

Canadian Limitations Notice

Notice: The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operational and safety requirements. The Department does not guarantee the equipment will operate to the user's satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local telecommunications company. The equipment must also be installed using an acceptable method of connection. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

Caution: Users should not attempt to make such connections themselves, but should contact the appropriate electric inspection authority, or electrician, as appropriate.

WEEE Statement

(Waste Electrical and Electronic Equipment)

July, 2005

The WEEE directive places an obligation on EU-based manufacturers, distributors, retailers and importers to take-back electronics products at the end of their useful life. A sister Directive, ROHS (Restriction of Hazardous Substances) compliments the WEEE Directive by banning the presence of specific hazardous substances in the products at the design phase. The WEEE Directive covers all Multi-Tech products imported into the EU as of August 13, 2005. EU-based manufacturers, distributors, retailers and importers are obliged to finance the costs of recovery from municipal collection points, reuse, and recycling of specified percentages per the WEEE requirements.

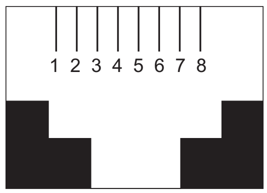
Instructions for Disposal of WEEE by Users in the European Union
The symbol shown below is on the product or on its packaging, which indicates that this product must not be disposed of with other waste. Instead, it is the user's responsibility to dispose of their waste equipment by handing it over to a designated collection point for the recycling of waste electrical and electronic equipment. The separate collection and recycling of your waste equipment at the time of disposal will help to conserve natural resources and ensure that it is recycled in a manner that protects human health and the environment. For more information about where you can drop off your waste equipment for recycling, please contact your local city office, your household waste disposal service or where you purchased the product.

Appendix A: Cable Pinouts

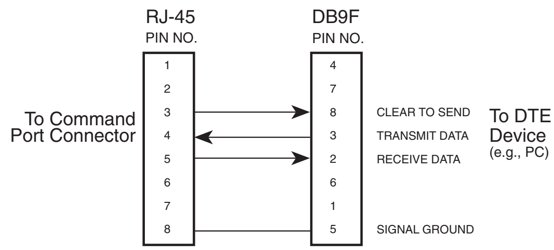
Appendix A: Cable Pinouts

Command Cable

RJ-45 Connector



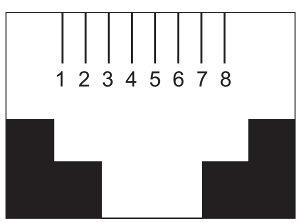
End-to-End Pin Info



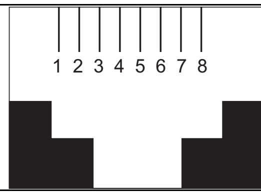
RJ-45 connector plugs into Command Port of MultiVOIP.
DB-9 connector plugs into serial port of command PC (which runs MultiVOIP configuration software).

Ethernet Connector

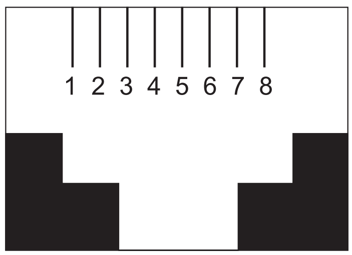
The functions of the individual conductors of the MultiVOIP's Ethernet port are shown on a pin-by-pin basis below.

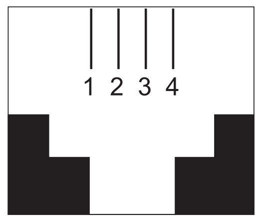
RJ-45 Ethernet Connector	Pin	Circuit Signal Name
	1 2 3 6	TD+ Data Transmit Positive TD- Data Transmit Negative RD+ Data Receive Positive RD- Data Receive Negative

T1/E1 Connector

T1/E1 Connector	
	1 2 } Receive Pair (from line) 4 5 } Transmit Pair (to line)

Voice/Fax Channel Connectors





Pin Functions (E&M Interface)		
Pin	**Descr**	**Function**
1	M	Input
2	E	Output
3	T1	4-Wire Output
4	R	4-Wire Input, 2-Wire Input
5	T	4-Wire Input, 2-Wire Input
6	R1	4-Wire Output
7	SG	Signal Ground (Output)
8	SB	Signal Battery (Output)

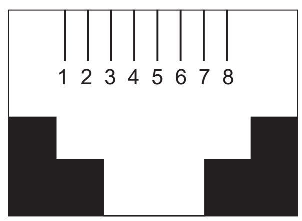
Pin Functions (FXS/FXO Interface)			
FXS Pin	**Description**	**FXO Pin**	**Description**
2	N/C	2	N/C
3	Ring	3	Tip
4	Tip	4	Ring
5	N/C	5	N/C

ISDN BRI RJ-45 Pinout Information

The S/T interface uses an 8-conductor modular cable terminated with an 8-pin RJ-45 plug. An 8-pin RJ-45 jack located on the terminal is used to connect the terminal to the DSL (Digital Subscriber Loops) using this modular cable.

The table below shows the Pin Number, Terminal Pin Signal Name and Network Pin Signal name for the S/T interface.

Pin	TE Signal		NT Signal	Pin
1	Not used		Not used	1
2	Not used		Not used	2
3	Tx+		Rx+	3
4	Rx-		Tx-	4
5	Rx+		Tx+	5
6	Tx-		Rx-	6
7	Not used		Not used	7
8	Not used		Not used	8



TE=Terminal Equipment

NT=Network

ISDN Interfaces: "ST" and "U"

The MVP410ST and MVP810ST are ISDN-BRI voip units that use an S/T outlet interface. You will need an NT1 device to connect these units to any network equipment that has the "U" ISDN interface. In the UK, and in many European countries, the telco supplies an NT1 device for ISDN-BRI service.

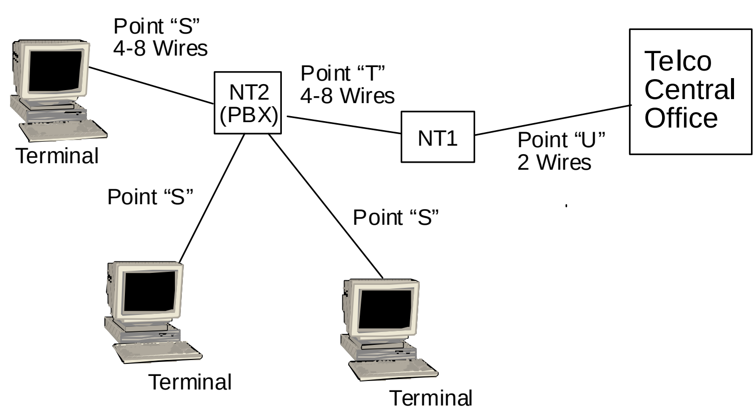
An ISDN Basic Rate (BRI) U-Loop consists of two conductors from the telco central office to the customer premises. The equipment on both sides of the U-loop accommodates the extensive length of the U-loop and the noisy environment in which it may operate. At the customer premises, the U-loop is terminated by an NT1 (network termination 1) device. An NT1 device makes an end-user's 4-wire terminal equipment compatible with the telco's 2-wire twisted pair ISDN-BRI line.

The NT1 drives an S/T bus. The S/T bus is usually made up of 4 wires, but in some cases may be 6 or 8 wires.

"S" and "T" refer to connection points in the ISDN specification.

When a PBX is present, *S* refers to the connection between the PBX and the terminal. ("Terminal" can mean any sort of end-user ISDN device: data terminals, telephones, FAX machines, voip units, etc.)

Point *T* refers to the connection between the NT1 device and customer supplied equipment. Terminals can connect directly to the NT1 device at point *T*, or there may be a PBX (private branch exchange, i.e., a customer-owned telephone exchange). The figure below shows "S" and "T" connection points in an ISDN network.



Appendix B: TCP/UDP Port Assignments

Well Known Port Numbers

The following description of port number assignments for Internet Protocol (IP) communication is taken from the Internet Assigned Numbers Authority (IANA) web site (www.iana.org).

> “The Well Known Ports are assigned by the IANA and on most systems can only be used by system (or root) processes or by programs executed by privileged users. Ports are used in the TCP [RFC793] to name the ends of logical connections which carry long term conversations. For the purpose of providing services to unknown callers, a service contact port is defined. This list specifies the port used by the server process as its contact port. The contact port is sometimes called the "well-known port". To the extent possible, these same port assignments are used with the UDP [RFC768]. The range for assigned ports managed by the IANA is 0-1023.”

Well-known port numbers especially pertinent to MultiVOIP operation are listed below.

Port Number Assignment List

Well-Known Port Numbers

Function	Port Number
telnet	23
tftp	69
snmp	161
snmp tray	162
gatekeeper registration	1719
H.323	1720
SIP	5060
SysLog	514

Appendix C: Installation Instructions for MVP428 Upgrade Card

Installation Instructions for MVP428 Upgrade Card

In this procedure, you will install an additional circuit board into the MVP410, converting it from a 4-channel voip to an 8-channel voip.

Summary: **(A) Attach four standoffs to main circuit card.**
(B) Mate the 60-pin connectors (male connector on main circuit card; female on upgrade card).
(C) Attach upgrade card to main circuit card (4 screws).

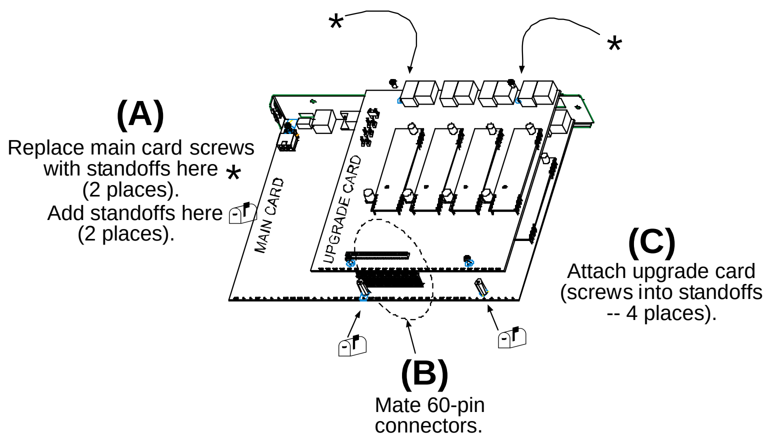


Figure D-1. Installation Summary

Procedure in Detail

1. Power down and unplug the MVP410 unit.

2. Using a Phillips driver, remove the blank cover plate at the rear of the MVP410 chassis. Save the screws.

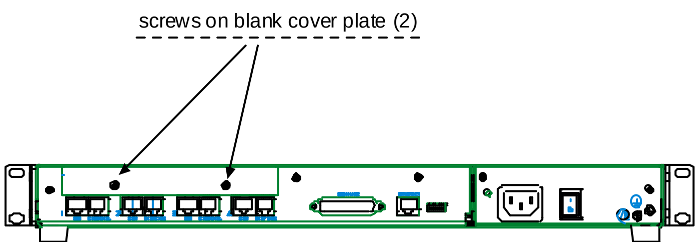


Figure C-2: Removing screws from blank cover plate

3. Using a Phillips driver, remove the three screws that secure the main circuit board and back panel assembly to the chassis.

> **NOTE:**
> Follow standard ESD precautions to protect the circuit board from static electricity damage.

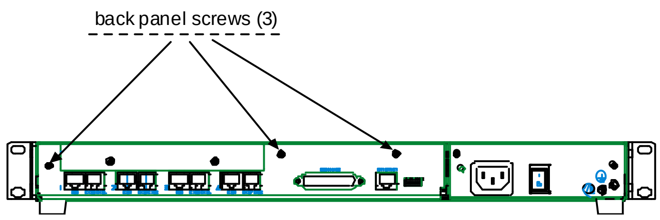


Figure C-3: Removing screws from back panel

4. Slide the main circuit board out of the chassis far enough to unplug the power connector.

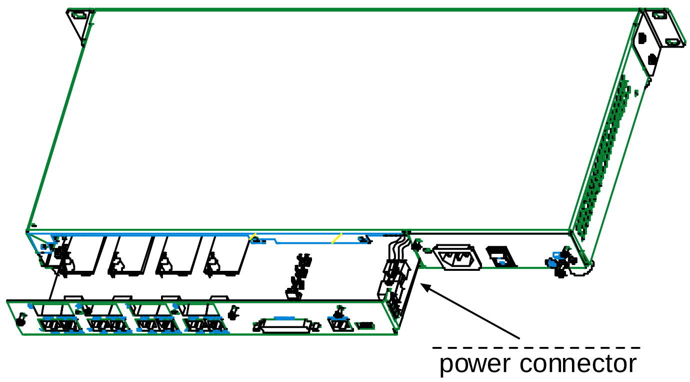


Figure C-4: Accessing power connector

5. Unplug the power connector from the main circuit board.
6. Slide the main circuit board completely out of the chassis and place on a non-conductive, static-safe tabletop surface.
7. Remove mounting hardware (2 screws, 2 nuts, and 4 standoffs) from its package.

8. On the phone-jack side of the circuit card, three screws attach the circuit card to the back panel. Two of these screws are adjacent to the four phone-jack pairs. Remove these two screws.

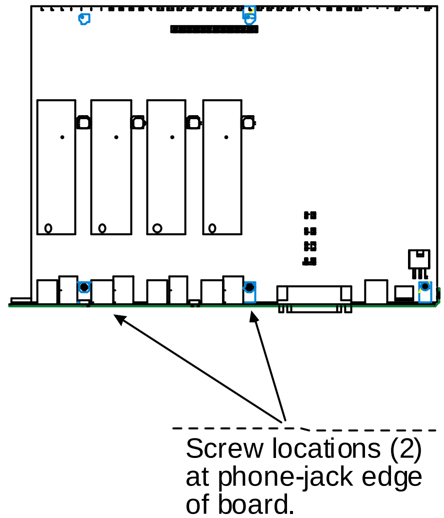


Figure C-5: Screws to be removed and replaced with standoffs (phone-jack edge of board; top view)

9. Replace these two screws with standoffs.

10. There are two copper-plated holes at the LED edge of the circuit card. Place a nut beneath each hole (lockwasher side should be in contact with board) and attach a standoff to each location).

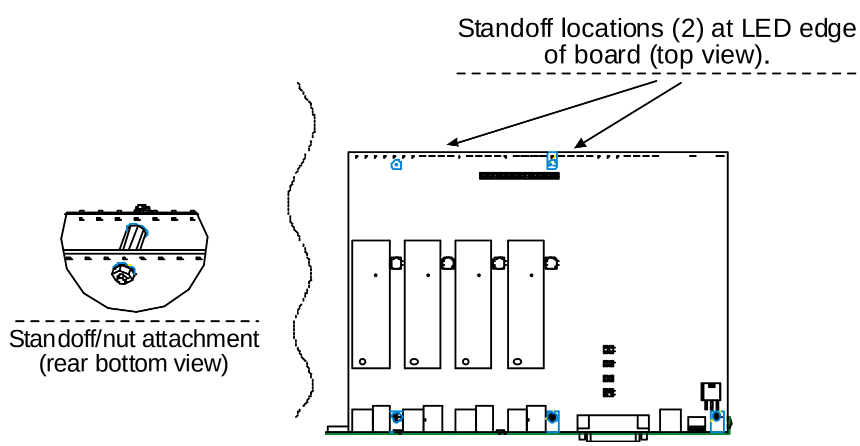


Figure C-6: Standoffs at LED edge of board (top view)

11. Locate the male 60-pin vertical connector near the LED edge of the main circuit card. Check that pins are straight and evenly spaced. If not, then correct for straightness and spacing. Locate the 60-pin female connector on the upgrade circuit card.
12. Set the upgrade circuit card on top of the main circuit card. Align the upgrade card's 4 pairs of phone-jacks with the 4 pairs of holes in the backplane of the main card. Slide the phone jacks into the holes.
13. Mate the upgrade card's 60-pin female connector with the main card's 60-pin male connector.

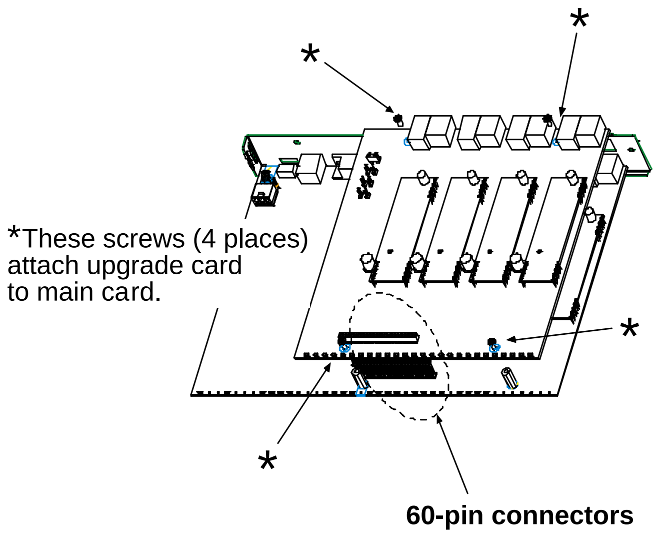


Figure C-7. Attaching upgrade card to main circuit card (secure 4 Phillips screws; mate 60-pin connectors)

14. There are four copper-plated attachment holes, two each at the front and rear edges of the upgrade card. Attach the upgrade card to the main card using 4 Phillips screws. The upgrade card should now be firmly attached to the main card.
15. Slide the main circuit card back into the chassis far enough to allow re-connection of power cable.
16. Re-connect power cable.
17. Slide the main circuit card fully into the chassis.
18. Re-attach the backplane of the main circuit card to the chassis with 3 screws.

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